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# Twelfth Symposium on Signal and Image Processing

## English-Language Abstracts

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# NAVAL OCEAN SYSTEMS CENTER

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# PREFACE

During the week of 12–16 June 1981, the Douzième Colloque sur le Traitement du Signal et des Images (Twelfth Symposium on Signal and Image Processing) was held in Juan-les-Pins, France. This symposium, sponsored by the Groupe de Recherche et d'Etudes du Traitement du Signal (Signal Processing Research Association), known as GRETSI, is held biennially in the spring, and for many years has been the foremost showplace for French work in all areas of signal-processing research, and particularly in underwater acoustics.

The Twelfth Symposium Proceedings are published in two volumes and are available from the following address:

Association GRETSI
7, chemin des Presses
Boite Postale 85
06801 Cagnes-sur-Mer CEDEX
France

Papers are published primarily in French, but to encourage international participation are also accepted in English. Sixty-four of the 255 papers in this year's proceedings are in English. All papers include English abstracts or summaries, which are gathered together and republished in the present document in the hope that they will be useful in promoting interaction between French and English-speaking researchers in the underwater acoustics and signal-processing field.

## CONTENTS

# Topic 1
# SIGNAL THEORY

## ANALYSE PAR ONDELETTES DES SIGNAUX ASYMPTOTIQUES: EMPLOI DE LA PHASE STATIONNAIRE

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### SUMMARY

The wavelet representation describes finite energy signals with scaled and delayed copies of a standard wave. It may be interpreted as a cross ambiguity function. For asymptotic signals the representation exhibits a main ridge which defines the so called "skeleton". It enables a rather accurate reconstruction of modulated signals. These properties appear very useful in the study of animal SONAR systems.

## ONDELETTES, SPECTROGRAMMES ET LISSAGES DE LA DISTRIBUTION DE WIGNER-VILLE

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### SUMMARY

Replacing the (energetic) wavelet transform in the general framework of time-frequency representations, it can be shown that, in similarity with the spectrogram, it results from a joint smoothing of some distributions which are better localized in the time-frequency plane, and whose prototype is the Wigner-Ville distribution. In contrast with the spectrogram, the smoothing associated to wavelets is not homogeneous over the plane: the transform is, in the frequency domain, of the form of a bank of constant-Q filters. Conditions for the smoothing relationships to hold are provided and the compared behavior of the three main tools (wavelets, spectrogram and Wigner-Ville) is illustrated by means of two typical examples.

## ON INSTANTANEOUS FREQUENCY AND CYCLOSTATIONARITY

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### SUMMARY

In this paper we derive expressions for the moments of instantaneous frequency of cyclostationary random signal. Our results show that cyclostationary signal is not only characterized by the periodicity of its statistical moments but, can be also characterized by the periodicity of the moments of its instantaneous frequency. This characterization appears to be valid for either single or multi-periodicity situation, in which case the first moment of instantaneous frequency can be represented in an $\alpha$-cyclic decomposed form. To prove this general property we also derive expressions for the higher-order moments of instantaneous frequency and discuss the necessary condition for their existence.

## SUR LA DUALITE ENTRE LA PREDICTION LINEAIRE D'UN SIGNAL ET LA DECOMPOSITION QR

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### SUMMARY

The triangular factorization of a signal data matrix called QR decomposition is related to linear prediction. In algorithms for adaptative filtering the factorization is carried out recursively through a set of rotations. In this paper relationships are established between the rotation parameters and the rotated data vectors and the reflection coefficients which appear in normalized lattice structures for linear prediction. The application to real time signal analysis is also considered.

## ESTIMATION DE LA STRUCTURE D'UN PROCESSUS STOCHASTIQUE MULTIVARIABLE

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### SUMMARY

In order to estimate in an efficient way a parametric model for a vector stochastic process, one has to specify not only the order of the model but also its structure. We present a procedure allowing to estimate the structure of a state-space representation of a multivariate stochastic process from measured output data. It is assumed that the observed time series is a realization of a process with rational spectrum or the output of a stable, invariant linear system driven by white noise.

We propose an algorithm which selects a maximal set of linearly independent rows of the Hankel matrix built upon the estimated covariance sequence. This set is obtained by sequentially testing the smallest singular value of submatrices of the Hankel matrix and yields estimates of the observability Kronecker invariants. The scheme when compared to the existing ones on simulated examples, exhibits better or at least comparable results for a much lower computational complexity.

## COMPRESSION D'IMPULSION EN LARGE BANDE

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### SUMMARY

In broad-band situations the true time-compression and delay radar ambiguity function must be substituted for Woodward's approximation. Some non classical properties of this function are presented as consequences of previous studies relative to the affine time-frequency representations of signals. In particular a fast computation technique is proposed which exhibits the same efficiency for broad-band plotting as the FFT in the narrow-band case.

## PULSE COMPRESSION WITH PERIODICAL BINARY PHASED SIGNALS

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### SUMMARY

In digital communication, navigation, and radar applications and for syncronisation and range measurement methods binary phase coded sequences are used with the property that their periodic or aperiodic auto-correlation function (acf) has an impulse-like form /1,2,4/. The concept of acf analysis implies a system receiver with matched-filter technique in practice /3/. In the case of binary sequences and from a signal processing point of view it is not necessary that the receiver filter coefficients are binary. Therefore, in this paper a mismatched-filter design is introduced where the reciver filter coefficients are used as free parameters which are optimized with the property that all sidelobes of the cross-correlation function (ccf) are zero. The present paper gives some results for binary sequences and the belonging mismatched-filters. For each length $N > 2$ exists at least one sequence and a belonging mismatched-filter which has no sidelobes at all in the output signal.

## DISTRIBUTION STATISTIQUE DE L'INTENSITE DU BRUIT DE TRAFIC ET DE L'AMPLITUDE DE LA REVERBERATION DE FOND: DEUX PROBLEMES MATHEMATIQUES VOISINS

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### SUMMARY

The background noise due to ships traffic or the bottom reverberation received by an active sonar both result from the summation of a large number of elementary components. From the mathematical point of view, these components are real positive terms (elementary noises intensities) for traffic noise, and complex terms (amplitude and phase of the elementary echoes) for reverberation. The transformations allowing the distribution of noise levels or reverberation amplitude to be computed from the statistical laws of the elementary components are presented. It is shown that, when the above distribution is not gaussian, a limiting value can be found above which the noise or the reverberation is principally due to a single component dominating the sum of all the other ones. Such a property can be used to define a criterion for the detection and estimation of a dominating noise component in traffic noise or a dominating echo in reverberation.

# THÉORÈME SUR LES SIGNAUX ANALYTIQUES

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## SUMMARY

The object of this paper is to present a theorem which allows the exploitation of the Bedrosian theorem. The Bedrosian theorem states that for any given two functions $f$ and $g$ with their frequency support of $F$ and $G$ (the Fourier transforms of $f$ and $g$ respectively), that satisfy certain conditions, the Hilbert transform of the product $f.g$ may be, reexpressed as $f.H_t[g]$.

The theorem proposed in this paper relieves the conditions of the Bedrosian theorem for a class of function and allows the equality $H_t[f.g] = f.H_t[g]$ even in the case where the support of $F$ and $G$ are not fulfilled. The advantage of this theorem is the enlargement of the class of signals to which the separation applies. On the other hand, an alternative proof of the Bedrosian theorem, based on the analytic form of the frequency supports, is proposed.

# Topic 2
# DIGITAL PROCESSING I

## UNE PRÉSENTATION UNIFIÉE DE FILTRAGE RAPIDE FOURNISSANT TOUS LES INTERMÉDIAIRES ENTRE TRAITEMENTS TEMPORELS ET FRÉQUENTIELS

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### SUMMARY

In this paper, we show that all the main types of fast FIR filtering algorithms can be derived as follows: decomposition of the signals and of the filter (either polyphase, or blockwisely), "diagonalisation" of the resulting pseudocirculant matrix, and reconstruction of the output by use of the Chinese Remainder Theorem. This presentation is shown to provide new possibilities. An application to adaptive filtering is also given.

## COMPUTING FREQUENCY TRANSFORMATIONS FOR LATTICE DIGITAL ALL-PASS FILTERS

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### SUMMARY

A numerical procedure is described for computing spectral transformations of all-pass filters realized in lattice form. The procedure can be applied, for example, to transfer functions implemented as the parallel connection of two all-pass filters, thus yielding a tunable filter realization. The method exploits the orthogonal Hessenburg structure of the state space description of a lattice filter, and the algorithm so obtained exhibits excellent numerical stability.

## ÉLIMINATION DES SINGULARITÉS DANS LE TEST DE SCHUR-COHN

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### SUMMARY

The Schur-Cohn algorithm allows us to determine the number $n_e(P)$ of zeros appearing outside the unit circle for a polynomial $P$. This algorithm is based on the computation of a set of $n$ coefficients $\rho_n, \rho_{n-1}, \ldots, \rho_1$ from $P$ assumed degree $n$. But there are polynomials $P$, referred to as *singular*, for which the computation of the $\rho_i$s cannot be carried out until the last coefficients $\rho_1$.

In this paper, we give a new version of the Schur-Cohn algorithm. This version allows us to compute from any polynomial $P$, even if it is singular, a set of $n$ coefficients $k_i$. It is shown that the number $n_e(P)$ equals the number of the $k_i$s with modulus greater than one. The new version of the algorithm is presented and illustrated on some particular polynomials and transposed for studying the zero location with respect to the imaginary axis.

## STABILITÉ D'UN FILTRE DE PRÉDICTION AVEC UN QUANTIFICATEUR DANS LA BOUCLE

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### SUMMARY

In this paper the stability of a transverse predictor is studied, taking into account a quantizer in the prediction loop. This loop introduces a recursive part in the prediction filter. We restrict our scope to an order 1 predictor. It is shown that the condition $|a| < 1$ for its coefficient is sufficient for stability, but is not necessary, which is an unexpected result. When $a = 1$ the system is the classical Delta modulation. The system instability is due to the quantizer saturation error. Two cases are studied. On the one hand the weak nonlinearity, where the quantizer is a saturation analytical function with a slope at the origin equal to one. On the other hand the strong nonlinearity, where the quantizer is an ideal relay. For a narrowband input signal, the stability limit is found close to 4 for a weak nonlinearity and equal to 2 for a strong one. For stability to hold it is required that the input amplitude is not too high w.r.t. the saturation level.

## ETUDE DES EFFETS DE QUANTIFICATION DANS LES ALGORITHMES MCR NUMERIQUEMENT STABLES

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### SUMMARY

We analyze the effects of the quantization of signals and internal variables on the stability and on the numerical precision of fast recursive least squares numerically stable algorithms (noted MCR) for adaptive transversal filtering. We first use simulations to compare the performances of the numerically stable MCR algorithms in fixed point and in floating point arithmetic, in the context of acoustic echo cancellation. We show that those algorithms keep the stability property when they are implemented in fixed point, and that the degradation of the mean square output filtering error results mainly from the quantization of the transveral filtering part in the algorithm. Then using a statistical approach, we present a theoretical analysis of the quantization effects in the filtering part, assuming that the errors due to the quantization of the Kalman gain are small in comparison with those due to the quantization of the tranversal filter.

## SUR L'EMPLOI DES ÉQUATIONS DE CHANDRASEKHAR POUR LA FACTORISATION RAPIDE D'UNE MATRICE PROCHE-DE-TŒPLITZ

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### SUMMARY

Fast inversion of close-to-Tœplitz matrices is a problem that frequently occurs in signal processing. The generalized Levinson, Cybenko or Schur algorithms are well known solutions to this problem. We show that a different solution, based on Chandrasekhar factorization, may be inbedded in the same vector recursion involving a $J$-orthogonal transformation. These new recursions produce an interesting way to compute the generators of a close-to-Tœplitz matrix. A particular attention is devoted to the parallelization of the resulting algorithm.

## A FAST PROBABILISTIC MEDIAN ALGORITHM FOR INTEGER ARITHMETICS

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### SUMMARY

In this paper, we introduce a new method for computing the moving median efficiently. The worst case complexity of the proposed moving median filter is of order $O(L)$, i.e. linearly proportional to the wordlength $L$. The complexity of the moving median does not depend on the number of samples included in the median computation. Also, the number of memory locations required for computing the median is low being at worst of order $2^{L+1}$. Furthermore, the control structure of the new median filter is very simple and it is easily implementable in VLSI. The algorithm is based on the multiresolution estimation of the moving histogram. Thus, the proposed algorithm can be used to find various percentiles and to estimate modes and other parameters of the data. Because the size of the moving window can be arbitrary long, the variance of the estimates to the percentiles can be made low. The basic algorithm given in the paper computes the median and other percentiles exactly but the algorithm can be also used to compute approximations to the true values.

## CLASSIFICATION AND ANALYSIS OF DIGITAL MODULATION SCHEMES USING WIGNER-VILLE DISTRIBUTION

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### SUMMARY

The Wigner-Ville Distribution has been investigated as a tool for the classification and analysis of digital modulation schemes. Starting from the complex baseband representation of modulations, the WVD of PAM and CPFSK have been derived. The time-frequency reliefs of digitally modulated signals show strong dependencies on the type of modulation and upon the modulation parameters for a specific type. These reliefs have been further parametrized through Haar functions and it has been found out that a few Haar coefficients provide significant features for the recognition of the type of modulation.

# THE NEWLY IMPLEMENTED SUPER PCM SYSTEM

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## SUMMARY

The concept of using abstract samples to modify the conventional sampling theory in a wide sense is introduced in this paper. The theory to be discussed here is sometimes called "WAVEFORM ABSTRACT THEORY [1]", which is a reasonable and powerful replacement of the conventional sampling theory. The WAVEFORM ABSTRACT is one of the noble keyes to the information era. Many digital communications problems can thus be solved or improved easily. For example, using the principles of WAVEFORM ABSTRACT to do Fourier Transform digitally is much simpler and far better than the so-called DFT/FFT methods. The theme of this paper is to present the fruitful result of the implementation of the Super-Pulse-Code-Modulation (SPCM) [2]. If this system is used for digital voice transmission, a 24K bits/s system of SPCM is much better than the 32k bits/s system of ADPCM. For music reproduction applications, it only takes 20k abstract samples to have nicer performance than a Compact Disk (CD) player. In this paper a description of the following three items will be given.

1. An introduction of SPCM and a comparison with PCM system.
2. The theory of Waveform Abstract (WA) and its coding.
3. The Super PCM system and its performance.

# FAST LEAST SQUARES SOLUTION OF CONFLUENT VANDERMONDE SYSTEMS OF EQUATIONS

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## SUMMARY

In this paper we introduce a fast algorithm for computing the QR factors of a complex column confluent Vandermonde matrix $V$. The complexity of the algorithm is $O(mn)$ where $m$ is the number of rows in $V$ and $n$ is the number of columns (we assume that $m > n$). This result generalizes a previous result presented in [9]. The matrices Q and R may be computed independently if desired. Such an algorithm allows for an important saving when solving systems involving such matrices in the least square sense, as for example when estimating the magnitude and phase of damped exponentials in the least square version of the Prony method, and some modes are repeated modes. Then the equation $V\underline{x} = \underline{y}$ is solved using $R\underline{x} = Q^*\underline{y}$.

## EFFICIENT NTT ALGORITHMS FOR PRIME NUMBERS

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### SUMMARY

In the paper the construction of Rader's number theoretic transform algorithms is described. It is shown that for $N=p$ being non-Fermat prime numbers the algorithms require significantly less operations than the known ones. Namely, the number of operations is reduced from $O(pd_1d_2...d_t)$ to $O(p(d_1+d_2+...d_t))$, where $d_i$ are mutually prime divisors of p-1. If applied to Mersenne transforms the approach results in algorithms which computational complexities are not higher than those for other NTTs, e.g. pseudo-Fermat ones. In general, the method can be used for improving small-N NTT modules in FFT-like algorithms. The paper contains also some general remarks on the transformation of DFT algorithms into those for number theoretic transforms.

## REALIZATION OF QUASI-CONTINUOUS DIGITAL FILTERS

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### SUMMARY

Quasi-Continuous Digital Filters (QCDFs) are digital systems the structure of which can be derived from analog filters. They are characterized by a quasi-continuous representation of signals, due to the pulse frequency modulation of the variables. Up-down counters and rate multipliers are the basic operators of this class of filters. These operators allow the synthesis of leap-frog filters, similar to RC active filters or switched capacitor filters. The round off noise resulting from the state variable quantization can be analyzed by means of usual software, using a classical digital filter structurally similar to the QCDF. This analysis determines the required number of bits for the up-down counters and the rate multipliers. The actual operators are realized on a VLSI integrated circuit by the mere abutment of the required number of bit slices and assembled to build up the QCDF.

## EXPLOITATION DE L'AMPLITUDE ET DE LA PHASE POUR LE TRAITEMENT D'ECHOS ULTRASONORES

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### SUMMARY

Physical band-limited signals picked up on sensors, always reveal phase variations which don't appear in the envelope. In this work, we use the concept of complex envelope which shows simultaneously envelope and phase informations versus time.

Then, we introduce complex envelopes computed on signals elaborated from analytical expressions. In the case of amplitude and frequency modulation, we show the advantages of such a representation. Finally, ultrasound echo signals are presented. The complex envelope really points out weak phase variations versus time and also represents simultaneously the evolution of the phase versus the amplitude in a polar plot.

## STRUCTURE DE FILTRES A ARITHMÉTIQUE DISTRIBUÉE POUR UNE OPTIMISATION EN TERME DE BRUIT NUMÉRIQUE.

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### SUMMARY

A new digital filter structure is proposed using distributed arithmetic. From a publication of F.J. Taylor (IEEE Transactions on Acoustics, Speech and Signal processing, vol. ASSP-34, October 1986) who suggests a digital filter realization with distributed arithmetic for a particular state space representation, the authors give a solution for a new structure usable with any state space representation. An error analysis for this structure is reported and verified by simulations which demonstrates the superiority of the new structure over traditional lumped parameter filters. But the major advantage of this realization is the possibility to minimize roundoff error effects without structure complications.

## CONCEPTION DE FILTRES NUMÉRIQUES HYBRIDES RIF/RII À PHASE LINÉAIRE

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### SUMMARY

A design method for hybrid FIR/IIR digital filters is proposed and evaluated. Given magnitude and linear phase characteristics are simultaneously approximated in the frequency domain using linear programming. The IIR and FIR components of the hybrid filter are globally considered.

In contrast with the previous work by Campbell et al, the proposed design is not restricted to the search for the best second-order denominator hybrid filter. To obtain the best hybrid filter design both the numerator coefficients and the denominator coefficients are simultaneously varied. We show that the best hybrid design is not always a second-order denominator, and under the same specifications we obtain better hybrid filters requiring a number of coefficients reduced by 25% to 30%.

The hybrid FIR/IIR design represents a trade-off between an FIR and IIR design: the resulting designs are compared with standard filters (Butterworth, Chebyshev, Cauer elliptic, optimal FIR). The various designs are compared on the basis of the number of storage locations for state variables and the group delay properties.

## UN ALGORITHME D'ÉLIMINATION DES TERMES D'INTERFÉRENCES DE LA TRANSFORMATION DE WIGNER-VILLE DESCRÈTE

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### SUMMARY

The qualities of signal analysis by the Wigner-Ville transformation are nowadays well recognized. Meanwhile, we recall its incompatibility with the sum of signals, and where do the cross-terms come from. The classical methods of elimination of the cross terms are shown, and we propose then a new one, based on image processing.

## MEDIAN FILTER: FREQUENCY DOMAIN ANALYSIS AND MEDIAN TYPE AVERAGING FILTER

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### SUMMARY

Frequency domain analysis has been presented for a class of signals. It is shown that the median filter acts as a spectrum subtractor. Certain interesting properties of the DFT ratios of input and output of the median filter are given.

For image processing applications it is shown that a modification to the structure of the running median filter, results in a better performance like noise removal, fast convergence and edge preservation. The performance of the average type filter is shown to be useful for these applications.

## ALGORITHMES DE FILTRAGE DE PETITE LONGUEUR ET LEUR UTILISATION EN FILTRAGE NON-RECURSIF

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### SUMMARY

This paper provides the basic tools required for an efficient use of the recently proposed fast FIR algorithms. These algorithms allow not only to reduce the arithmetic complexity but also maintain partially the multiply-accumulate structure, thus resulting in efficient implementation.

A set of basic algorithms is derived, together with some rules for combining them. Their efficiency is compared with that of classical schemes in the case of three different criteria, corresponding to various types of implementation. It is shown that this class of algorithms (which includes classical ones as special cases) allows to find the best tradeoff corresponding to any criterion.

# Topic 3
# DETECTION: ESTIMATION

## SUR L'INTERET DE L'ADAPTATIVITE PAR RAPPORT A LA ROBUSTESSE EN DETECTION SONAR

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### SUMMARY

This paper is devoted to the comparison between three receivers, two being minimax robust (the matched filter and the soft-limitor) and one being adaptive, i.e., based upon a parametric modeling whose parameters values are computed from the observation. This model is associated with the noise probability density function and corresponds to a Gaussian-Gaussian mixture PDF.

This comparison is made by using the ROC curves as detection criterion and with real underwater noise samples. For high signal-to-noise ratio, the adaptive receiver leads to an important improvement of the performances, whereas at low SNR, the gain is smaller.

## SOME RESULTS ON NONGAUSSIAN SIGNAL DETECTION

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### SUMMARY

Recent results on detection of random nonGaussian signals in additive Gaussian noise are discussed. A discrete-time approximation to a likelihood ratio detection algorithm is discussed, and preliminary results are presented for a computational evaluation of the performance for this approximation.

## MAXIMUM A POSTERIORI LIKELIHOOD DETECTION USING LEMPEL-ZIV ALGORITHM

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### SUMMARY

A semi-deterministic scheme is presented for detecting a deterministic signal in white Gaussian noise and a random transient disturbance. This scheme estimates the disturbance via the maximum a posteriori likelihood method and then forms the likelihood ratio using this estimate. The estimation proceeds in two stages: first, presence or absence of the disturbance is decided, and, if present, then the disturbance itself is estimated. Two examples are given to illustrate the scheme: in the first, the disturbance has a known shape with a random appearance time; in the second, it is a Gaussian noise with a partially Rayleigh distributed amplitude. The Lempel-Ziv algorithm is proposed as a simpler alternative to the computationally burdensome procedure for deciding the presence of the disturbance. Such an alternative is advantageous when the disturbance is rare and, hence, the matched filter is appropriate most of the time.

## PREDICTION LINEAIRE-QUADRATIQUE, ETUDE D'UNE RECURSIVITE SUR L'ORDRE, EXTENSION DE L'ALGORITHME DE LEVINSON

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### SUMMARY

The problem of predicting a sample drawn from a discrete real valued zero mean process, with finite moments up to the fourth order, using a discrete Volterra linear-quadratic filtering of its past is considered. The derivation is based on a completely new framework, based on linear-quadratic spaces, which takes into account properly the contribution of quadratic observations and their redundancy. Relying on this framework, we present the extended Normal Equation in a fashion very close to the linear. We show how the third order moments introduce a coupling between linear and quadratic terms. Moreover, we justify how we always improve the performance of a linear predictor by adding a quadratic filtering as soon as the underlying process is non gaussian. In keeping with this idea, we give an interesting interpretation, as a quadratic form, of the amount by which the improvement is gained. In order to derive an order recursive solution of linear-quadratic problem, we extend the classical Levinson algorithm, in the case of a stationary process up to the fourth order. As a matter of fact, because of the contribution of quadratic observations, we still obtain an orthogonalization procedure but with new features related to the Chandrasekhar equations and multidimensional lattice filtering.

# ESTIMATION NON LINÉAIRE DE L'AMPLITUDE D'UN SIGNAL

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## SUMMARY

The problem of estimating the amplitude of a signal appears in many aspects of signal processing such as, for example, amplitude modulation in communication theory. The calculation of the maximum likelihood estimator is often impossible because of lack of knowledge concerning the probability distribution of the noise. It is sometimes sufficient to use a linear estimator without bias and with minimum variance. Its calculation needs only knowledge of the correlation matrix of the noise. In this paper we want to show that it is possible to use a non linear estimator without bias and with minimum variance as soon as some higher order moments of the noise are known. Using the structure of a Volterra filter we give the equations allowing the determination of the optimum filter. These equations are solved in some particular cases in order to show the interest of the method.

## DETECTION USING CROSS-TERMS IN THE WIGNER-VILLE DISTRIBUTION

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### SUMMARY

Using the Wigner-Ville distribution (WVD) for general purpose time-frequency (TF) analysis of nonstationary signals has a lot of advantages over the other methods but has also one very important drawback: generation of cross-terms between different signal components which makes the interpretation of the spectrum difficult in many cases. A goal of this paper is to emphasize the fact that the main weakness of the WVD is profitable in detection applications.

The paper briefly reviews the cross-terms generation mechanism of the WVD, recapitulates the TF formulation of optimum detection by means of the WVD and puts forward an idea of new detection scheme that directly take advantage of the cross-terms generation in the WVD. In the proposed method a test signal is computed. Since the cross-terms always lies symmetrically in the mixed TF plane between objects which generate them, knowing the TF localization of the test signal and its interferences with all unknown components of analyzed signal we can find TF mixed time-frequency signal representations (MTFRs) modulation laws of these components using simple geometrical rules.

In comparison with standard techniques the presented idea realizes the detection of signal components in indirect ways. In the paper the method is described in a more detailed way and initial experimental examples of its using are given.

The proposed method can be especially useful for additional verification of hypothesis.

## SEPARATION DE MELANGES DE SIGNAUX

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### SUMMARY

The problem consists of isolating p independent stochastic processes from the observation of an unknown linear transform of them. The algorithms designed in the sequel are solely based upon the assumption that original processes are statistically independent. Those unknown processes are called sources. For the sake of simplicity, the linear transform is assumed to be instantaneous, namely defined by a constant matrix, A. Moreover, A is assumed square and regular in order for the problem to be solvable up to a diagonal multiplicative matrix. Two algorithms are compared. We introduce both the original adaptive algorithm of Jutten and Herault proposed in 1985 whose limitations are emphasized, and a new one performing better. The latter algorithm aims at cancelling the output cross-cumulants.

## REGROUPEMENT DE PISTES ANGULAIRES ISSUES D'ANTENNES PASSIVES DISPERSEES

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### SUMMARY

A track to track association test for bearings measured by distant platforms is presented here. The binary test is first introduced (one track for each platform, each track issued (or not) from the same source). Its performance is given particularly in ambiguous cases due to unobservable situations. Then, we generalize this test in a multitarget environment on each array.

## ON THE TIME DELAY MEASUREMENTS BY THE AVERAGE MAGNITUDE DIFFERENCE FUNCTION

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### SUMMARY

It is known that the accuracy of the Average Magnitude Difference Function (AMDF) based time delay estimate approaches the one of the classical cross-correlation-based estimate with a lower computational effort. However, some of aspects concerning the sampling rate necessary to search the extremum of the AMDF are often neglected.

In this work, the interpolation problems are addressed and the theoretical values of the variance of the estimate are given for a reference case. For comparison purposes, the theoretical accuracy is also given under the same conditions for the cross-correlation based (Direct) estimate.

## IDENTIFICATION DES CHAINES DE MARKOV CACHEES: APPLICATIONS AUX CANAUX NON-LINEAIRES

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### SUMMARY

The aim of this paper, is the application of the BAUM and WELCH algorithm to the identification of a non linear channel with additive white noise. From these algorithms, we derive recursive algorithms for the estimation of the parameters of the transmission channel and for the estimation of the probability law of the source.

## IDENTIFICATION DE MODELES ARMA

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### SUMMARY

After a brief review of the main identification methods of ARMA models, some new algorithms are presented. Then the links which exist between these new algorithms and other already existing methods are discussed. Finally the performance of various methods of ARMA parameters estimation is compared through a Monte Carlo analysis realized from simulated examples.

## REDUCED ORDER MODELING OF HARMONIC SIGNALS

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### SUMMARY

In this paper the problem of estimating frequencies of noisy sinusoids when the exact model structure is not a priori known is considered. Two algorithms are proposed for simultaneous frequency and model order estimation. The first one, applicable off-line, is based on the generalized least-squares method, while the second one is recursive, based on the RML method. The signal model is supposed to be in the casacaded notch form. Order estimation is based on direct residual power tests. Experimental results illustrate the characteristic properties of the methods which represent simple and reliable tools for practice.

## DÉCONVOLUTION DE PROCESSUS IMPULSIONNELS AVEC CALCUL EXACT D'UN CRITÈRE MAP

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### SUMMARY

This article presents an iterative maximum *a posteriori* deconvolution method for Bernoulli-Gaussian processes. This detection-estimation problem is formulated as that of a change of initial conditions in a least-squares problem. This yields an algorithm with a very simple structure which allows the evaluation of the MAP criterion without any approximation. The resulting method is easy to implement, computationally inexpensive and remains nearly optimal.

## ESTIMATION DES DATES D'ARRIVEE DE TRAJETS DE PROPAGATION EN TOMOGRAPHIE OCEANIQUE

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### SUMMARY

For oceanic tomography, it is essential to well identify and estimate the arrival times of the acoustic paths. The classical method which uses large WT signals and correlations may be no longer sufficient to well separate and identify the different time delays. So some high time resolution methods are necessary. In this paper, we present a method which performs a multidimensional filtering on the successive correlation data, and, after that, elaborates a good delay separation by some kind of deconvolution. Some simulation results and real acoustic experiment results demonstrate the good performance of the method.

## FILTRE OPTIMAL DE DÉTECTION DE RUPTURES MARKOVIENNES

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### SUMMARY

Optimal filtering theory provides a solution for rupture detection in dynamic systems. In this paper we are interested in the performances of one such filter where jumps are modelled with a finite Markov chain. In particular we analyze the estimate consistence, the filter robustness and the detection delay and false alarm rate evaluation.

## ANALYTICAL EVALUATION OF THE PERFORMANCE OF THE SQUARE LAW RECEIVER FOR GAUSSIAN TARGETS IN GAUSSIAN CLUTTER

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### SUMMARY

The paper analyses the performance of Square Law Detectors in terms of Receiver Operating Characteristics, when both the signal and the noise are Gaussian stochastic processes.

The theoretical results valid for the optimum receiver are extended and applied to the case of some commonly used MTI filter.

An evaluation and design methodology is outlined.

## COMPARISON OF SIGNAL SEPARATION METHODS BASED ON SVD AND GSVD


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### SUMMARY

In this paper two signal processing principles, that try to separate signals from a set of recorded signals by finding the optimal linear combination of these data signals, are presented and compared. The methods are based on two mathematical concepts and on the computation of the (Generalized) Singular Value Decomposition. They are applied to the problems of FECG extraction from potential signals on the maternal skin and to speech enhancement.


## SIGNAL PROCESSING IN THE MIXED TIME-FREQUENCY DOMAIN USING SVD AND WIGNER-VILLE TRANSFORM TECHNIQUES


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### SUMMARY

The Wigner-Ville distribution (WVD) is a two dimensional time-frequency (TF) representation of a one dimensional time signal and it is obvious that there exists a redundancy of information contained in it. The paper discusses a problem of optimal application of the singular value decompostion (SVD) for the compression of the WV-spectrum. General conculsions are formulated in it and some examples of signal processing in the mixed TF domain by means of the WVD and the SVD are given. A short performance comparison is made between the presented technique and the other methods of signal processing in the case of noise filtering.

## CLASSIFICATION DES SIGNAUX MAGNETIQUES TBF SOUS CONTRAINTE DE DECISION

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### SUMMARY

This paper presents the development and the estimated performances of a new ULF magnetic signal analysis method allowing the possibility to classify the analysed wave under sooner decision constraint from the identification of a given set of parameters.

The signal is a three components vectorial signal which represents the magnetic field created by a moving ferromagnetic body and observed by a fixed sensor.

The bouy is assumed to be modelised either by a dipole or a uniformly magnetized ellipsoid.

The method described in this study is based on a known magnetic signal space of which has been built an orthonormal basis that makes possible to have an exact analytical generator of the dipolar or ellipsoidal signals.

The signal processing principle is built according to a projective method in this space. It lies in the comparison of the projection energy of the observed signals, including natural geomagnetic noise, over all possible basis related to the parameters to be identified.

After the description of the problem and the assumption we have made, we present the processing set and the simulation results from which we have estimated the theoretical performances of the method.

By another way, results from a statistical analysis which has been conducted with a large number of possible scenarios have allowed us to evaluate the effectiveness of the algorithm seen as a measure of the fitness of the method to classify the received waveforms.

## POURSUITE DE CIBLES MANOEUVRANTES PAR DES ALGORITHMES MARKOVIENS HYBRIDES

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### SUMMARY

In this paper, a comparison between various tracking algorithms for civil airplanes is presented. As compared to conventional tracking methods, these algorithms, based on hybrid Markov processes, allow a dramatic improvement on the tracking accuracy. Several dynamic models can be taken into account as well as sudden jumps between the different models. Moreover, they are not strictly limited to target tracking problems but can also be applied to any estimation problems in a switching environment (failure detection, chemical processes with time-dependent dynamics,...)

## INTERPOLATION DE SIGNAUX, APPLICATION A L'ESTIMATION DE SIGNAUX IMPULSIONNELS NOYES DANS UN BRUIT ADDITIF STATIONNAIRE A BANDE ETROITE

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### SUMMARY

The purpose of this paper is to present an adaptive interpolation method for stationary signals that can be modeled by an autoregressive process. Estimation of missing samples is achieved by minimizing residual error, which depends on the model coefficients.

This method ensures good results when applied to the estimation of impulsive signals in band limited stationary additive noise. It has been applied successfully to signal to noise ratio enhancement of signals obtained during Eddy currents non destructive testing.

## APPLICATION DU FILTRAGE NON LINÉAIRE EN TRAJECTOGRAPHIE PASSIVE

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### SUMMARY

We apply optimal nonlinear filtering techniques to a target motion analysis problem with bearings — only measurements. We present a rather simple example. We introduce a non-linear filter (which is a space discretization of the optimal nonlinear filter) and we compare it to the extended Kalman filter.

## DÉTECTION ET ESTIMATION POUR LA TRANSMISSION SUR UN CANAL INCONNU

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### SUMMARY

We present an iterative method for joint channel parameters estimation and symbol detection via the EM algorithm. Channel parameters, including noise variance, are estimated using maximum likelihood criterion. Minimum error probability decisions on symbols are easily obtained at the end of iterations. The proposed receiver is valid for both linear and nonlinear channels. It allows an improvement in the system throughput by sparing the usual transmission of known symbols employed for channel identification.

## INTERET DE LA DECOMPOSITION EN VALEUR SINGULIERE (SVD) EN TRAITEMENT DU SIGNAL SONAR

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### SUMMARY

In this paper, the Direct Data Approximation (DDA) of stochastic system identification is applied to the linear equal spaced array narrowband source direction finding problem. The power of state space parametrization and the numerical properties of the Singular Value Decomposition (SVD) based on realization algorithms together make the new method (the DDA method) very exciting. Simulations performed on the DDA algorithm demonstrate its numerical robustness compared to existing methods based on the covariance matrix analysis.

## DÉTECTION DES OBJETS FAIBLES DANS DES IMAGES CÉLESTES À L'AIDE DE LA TRANSFORMÉE ONDELETTE

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### SUMMARY

Deep astronomical images show a high number of extended objects, superimposed on a variable sky background. The classical detection method is based on the cross-correlation with stellar profile after subtraction of a background map. This procedure is only adapted to the detection of isolated faint stars.

*The Wavelet Transform* is a set of filtering with a function with a varying size. We used different analysing wavelets: mexican hat (Laplacian-Gaussian), the one resulting from a local gaussian fit, the one related to an algorithm "*à trous*".

Experimentations on astronomical images led to an efficient detection of objects with different size. They can be included the ones in each others, one object being only detected for a given scale range.

This analysis gives a very rational detecting procedure, which does not contain any arbitrary parameters, as the background scale or the size of the objects to be detected.

## DETECTION DU NOMBRE DE SOURCES PAR LA METHODE DES ESPERANCES CONDITIONNELLES, CALCUL PAR NOYAUX

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### SUMMARY

A very important problem in submarine detection is to estimate the precise number of sources. Each boat generates a random pressure field. After propagation in the supposed linear random medium the p waves plus additive noise are observed.

The number p of sources is the dimension of the vectorial space generated by the random expectations of the q observations conditionally to their arithmetic mean. This property can be used to obtain a non linear estimator and with elementary operations the number p was obtained.

But today, by kernel statistical technic, these conditional expectations are estimated from simulated or experimental data and an observation time very short is needed to obtain the same precision.

# DÉTECTION RÉPARTIE ET QUANTIFICATION

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## SUMMARY

In many distributed detection systems separated sensors and a central processor are used. Because of some constraints on communications all the information from the sensors cannot be sent to the central processor. This makes the perception of the system less accurate and induces a loss in performance. Previous studies have considered the case where a detection procedure is realized at each sensor. In this paper a new approach within the framework of quantization is presented. The two procedures are compared in Gaussian noise environment. It is shown that the quantized structure can reach better performance. The optimization of quantized systems according to the Neyman-Pearson criterion is shown to be difficult. Leaving the concept of absolute optimality, we determine the parameters of the quantized systems using heuristics arguments based on the geometrical interpretation of the decision domains.

# Topic 4
# SPECTRAL ANALYSIS

## THE ROLE OF PERIODOGRAM ENVELOPE IN SPECTRAL ESTIMATION

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### SUMMARY

The concept of spectral envelope derived from the given periodogram is introduced. The interest of this function is shown both in the non-parametric as well as in the parametric approach. The nice properties of the envelope associated to the magnitude of the so-called analytic spectrum, aprize the new function to be the most suitable candidate for additional constraints in variational spectral estimation. Finally, a well supported ARMA spectral estimate from correlation and envelope constraints is derived. In fact the use of periodogram envelope in spectral estimation, both for final estimates or for prior estimates in more complex procedures, should be prefered with respect to other possible functions like cepstrum. This is due to the smooth character, the statistical stability and the linear model, derived from the causal autocorrelation function, that the spectral or the periodogram envelope has.

## ESTIMATION DE LA MATRICE SPECTRALE DE SIGNAUX CERTAINS: APPLICATION À LA SÉPARATION D'ONDES EN SISMIQUE

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### SUMMARY

The separation of seismic waves by means of the spectral matrix can be simplified under certain conditions that we will set forth in this paper. After specifying the characteristics of the seismic signals at the time of matrix estimation, and the definition of a source, we will show the limits of separation on synthetic signals. On real signals we will show that it is possible in VSP (vertical seismic profile) [Mari and Coppens] to extract and separate rising, descending, and conversion waves.

## FAST NEWTON-RAPHSON METHOD FOR MOVING-AVERAGE SPECTRAL FACTORIZATION

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### SUMMARY

In this paper, we present an implementation of the Newton-Raphson method to compute the minimum phase moving-average spectral factor of a finite positive definite correlation sequence. The major advantage of such a method is contained in its quadratic convergence behavior. Each step in the successive approximation method involves a system of linear equations that is solved using either the Levinson algorithm backwards (the Jury stability test), or a symmetrized version of the Euclid algorithm. Various properties of the Newton-Raphson map are studied. The overall efficiency of the method is then compared with the classical methods of Durbin and Bauer.

## UNE MÉTHODE DE MAXIMUM D'ENTROPIE APPROCHÉE POUR L'ESTIMATION SPECTRALE 2D

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### SUMMARY

The problem of 2D Maximum Entropy Method (MEM) spectral estimation from a 2D set of finite correlations $\{r(m,n), |m|\leq M, |n|\leq N\}$, leads usually to a non linear problem which is computationally expensive. A new algorithm yielding approximate MEM spectral estimates is proposed, based on linear prediction methods. It rests on the relation giving the entropy of hybrid spectra with infinite order and, on the quadratic expansion of this relation with respect to the unknown correlations. The obtained spectra are very close to the true ME ones, for a much lighter computational burden.

# ESTIMATION SPECTRALE 2-D

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## SUMMARY

This paper deals with 2-dimensional spectral estimation, especially in two cases: estimation of plane waves angles-of-arrival and frequencies using a linear array of sensors (spatio-temporal spectral analysis) and bearing estimation in the 3-D space (2 angles are needed to determine each source) using a planar rectangular array. We show that if the sources are non totally correlated this problem can be solved using a 2-D generalization of Kung's method. Moreover, we show that the method requires only 2 sensors in one dimension, provided that the number of sensors in the other dimension is at least equal to the number of sources. If the sources are totally correlated, we present a method to retrieve the sources, without spatial smoothing.

# GENERALIZED WIENER-G FUNCTIONALS FOR NON-LINEAR SYSTEMS APPLICATION TO PARTIALLY COHERENT IMAGING

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## SUMMARY

We demonstrate that the imaging of a 3-D distribution of index of refraction by a partially coherent optical system involves the transmission of series of Volterra functionals of the object function, that represent either the mutual intensity or the image intensity. According to Wiener, an optimal description of this non-linear process is based on the definition and calculation of series of orthogonal functionals. When applied to partially coherent imagery this approach requires the introduction of a scalar product different from that used by Wiener for temporal signals. A set average that characterizes the spatial — or spectral — properties of a specific class of optical objects to be imaged is proposed. Up to the third moments of the object class are shown to be necessary to account for the transmission of 3-D informations in partially coherent imaging. They are supposed to be available to work out the functional orthogonalisation scheme. As a consequence, generalized Wiener G-functionals (GWGFs) are derived, that yield an orthogonal representation of the mutual spectral density, and the image spectral density. Similarly to Wiener's work on the class of white, Gaussian temporal signals, the successive kernels of the GWGFs have the property of being expressed in terms of the leading kernel of each GWGF. But the symmetry features of Wiener G-functionals do no longer hold here.

The next and necessary step in the implementation of GWGFs is their expression into series of orthogonal functions. At the level of modeling adopted in this first paper it is not useful to specify this basis. Orthogonal series expansions of the GWGFs of both the mutual spectral density and the image spectral density are provided. Within this framework the object is described as a vector, and the imaging system is characterized either by signal-independent matrices and tensors (for the transfer of the mutual spectral density) or by signal-independent vectors, matrices and tensors (for the transfer of the image spectral density).

## LOUPE PAR FILTRAGE FREQUENTIEL

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**SUMMARY**

In this paper, we study a new real time method of spectral resolution increase, called frequency filtering zoom (LFF). The DFT's use allow to execute the operations of frequency filtering and translation in base-band in the frequency domain. So, contrary to the complex demodulation zoom (ZFFT), this technic allow to save the informations corresponding to the signal power and phase.

## UNE METHODE D'ESTIMATION DE SPECTRES DISCRETS FONDEE SUR LE BATTEMENT DES FREQUENCES – QUELQUES PROPRIETES STATISTIQUES

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**SUMMARY**

We present a computationally simple and efficient method for estimating the line spectrum of a signal corrupted by a stationary ARMA noise. Then, a statistical study of asymptotical properties of the estimators yielded by this method is given, that shows that these estimators are asymptotically among the most accurate, since they have the same asymptotic variance as those obtained with Whittle's method.

## NON-UNIFORM FILTERBANK DESIGN USING THE RUNNING FOURIER TRANSFORM

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### SUMMARY

This paper describes a technique for designing uniform and non-uniform filterbanks based on the Running Fourier Transform (RFT). The RFT is implemented by convolving the input signal with one of a family of windows, $H(nT) = (nT)^k e^{-\alpha nT}$, where k and $\alpha$ may be chosen to specify the order and bandwidth, respectively, or each analysing filter. The RFT is superior to the more traditional Discrete Fourier Transform (DFT) in that non-uniform channel spacing with variable analysing bandwidth is permissible and, also, spectral leaking across channels can be controlled.

By viewing the operation of the RFT filterbank in terms of bandpass filtering, an expression for the equivalent composite impulse response has been derived. This response can be used to optimise the composite amplitude response of the RFT filterbank. Filterbanks with non-uniform channel spacing and bandwidth can be designed by splitting the frequency band of interest into a number of uniform sections and then optimising the equivalent composite impulse response of each section independently. Finally a modified cepstral smoothing technique for non-uniform spectra is presented and shown to be superior to conventional bi-pass filtering.

## ANALYSE DE CHAMPS TURBULENTS PAR TRANSFORMÉE EN ONDELETTE

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### SUMMARY

A signal analysis algorithm using periodical wavelet decomposition is presented. After a method restate, two examples of application in the one-dimensional case are presented: a fundamental example and a turbulent experimental signal.

# Topic 5
# COMMUNICATIONS

## SUR LES SCHEMAS DE CODES CONCATENES AVEC DECODAGES PONDERES

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### SUMMARY

Concatenation of two error-correcting codes is current practice. An inner code, often decoded using a weighted input, is concatenated with an outer code decoded without such a weighting. If, however, the output of the inner decoder itself is weighted, the error rate is improved. When the binary alphabet is used over a memoryless Gaussian channel, we propose a means for predicting the performance of the whole concatenated scheme. The part of the inner coding and the benefit resulting from its weighted-output decoding are emphasized.

## STRATÉGIES EFFICACES DE RETRANSMISSION UTILISANT LE CODAGE CONVOLUTIONNEL

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### SUMMARY

In this paper we present and analyse efficient ARQ schemes based on rate compatible punctured and repetition convolutional codes. The great advantage of these ARQ strategies is that the throughput efficiency with a starting high rate code, $R > 1/2$, is always better than with a starting rate $1/2$ code. This makes the use of high rate codes attractive, allowing thus the system to be adaptive to channel conditions, even under severe degradations and wide variations.

## MODELING OF ERRORS ISSUED FROM MAXIMUM LIKELIHOOD DECODING

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### SUMMARY

Maximum Likelihood (ML) decoding has been used in many cases, such as classical error control techniques, coded modulation techniques and partial response encoding techniques. The errors issued from this decoding occur in bursts and cannot be represented by the classical Binary-Symmetric-Channel. This article is dedicated to introducing an analytical method for modeling such errors. A Markov Model is used and is based on the ML decoding error producing mechanism. This model can be used to design and study "super coding" techniques such as concatenated or cascaded codes and it can also be used to regenerate a similar error sequence without actually encoding and decoding the signal, and this can save a lot of simulation time.

## MODULATIONS CODEES EN BLOCS BASEES SUR LA PARTITION-C DE L'ALPHABET

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### SUMMARY

This paper presents a class of block-coded modulation (BCM) schemes based on two-step partitioning of the signal constellation. They use the same alphabet expansion as Ungerboeck's trellis-coded modulation (TCM). Using short block lengths (3 and 4), specific modulations are described that transmit 4, 6, and 8 information bits per symbol. The coding gain over equivalent uncoded modulation is only 1.8 dB for block length 3, and 3 dB for block length 4, but their detection simplicity makes the presented BCM scheme very attractive for high-speed applications. In particular, they may be of potential interest to high-capacity digital microwave radio systems. The coding gain is also investigated on nonlinear channels and found to be significantly higher than on additive white Gaussian noise channels.

## UNE TECHNIQUE DE PREDISTORSION ADAPTATIVE A INTERPOLATION INTERSYMBOLE

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### SUMMARY

In this paper, we present a new data predistortion technique using intersymbol interpolation for efficient compensation of high-power amplifier (HPA) nonlinearities in digital microwave radio systems employing QAM signal formats. It is a $T/3$ – spaced data predistortion technique which cancels nonlinear distortion at the HPA output at three points per symbol interval. In addition to constellation warping, this technique also compensates for nonlinear intersymbol interference, and achieves a significant performance gain over conventional data predistortion. With respect to the $T/2$ – spaced predistortion technique presented by the present authors in [5], the present one uses narrower pulse shaping at the transmitter and achieves a higher protection against adjacent channel interference. Using the 64 and 256 QAM signal constellations, it is shown that the proposed technique leads to a very efficient utilization of the available HPA power.

## *MODULATIONS MAQ2$^N$ DE REDEMENT (N-1)/N*

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### SUMMARY

The use of error-correcting codes gives M-ary modulations more efficiency by increasing their noise immunity. Trellis coded modulation (TCM) family uses convolutional codes. The results of Ungerboeck approach are summarized. They concern the optimal design of coding by taking account of the modulation. The method used by Calderbank allows the global definition of families of coded modulations using the same convolutional code.

For the MAQ2$^N$ modulations with rate (N-1)/N, the maximum attainable coding gain is related to the trellis complexity, exactly to the number of states in the Viterbi decoder, and is nearly independent of the number of states of the modulation. 8QAM with 2/3 rate gives the performances and the optimal receiver structure for all the modulations of the family. This result shows the receiver can be implemented in two circuits, a preprocessor and Viterbi decoder, this structure being suited to the reception of all the modulations with different rate and using the same convolutional code.

## OPTIMUM STRUCTURE OF M-ARY MODULATED SIGNALS FOR ERROR DETECTING CODES

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### SUMMARY

In some ARQ schemes, the combination of modulation and channel coding improves the performance of a communication system. This paper describes a new method for the integration of an error detecting code with a M-ary modulation. The described method permits to improve both the error probability and the throughput of an ARQ protocol with respect to other similar schemes.

## SYSTEME DE TRANSMISSION NUMERIQUE A PARTAGE EN FREQUENCE

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### SUMMARY

In this article we describe a frequency domain modem which uses some new techniques to provide reliable high-speed data transmission over long-distance telephone lines. We show that frequency domain techniques provide important advantages compared to classical time domain techniques. The first part of this article describes the basic principles of frequency domain data transmission and their relationship to the line spectrum and impulse response. The second part consists of an over-all description of the modem's operating principles and of the frequency domain algorithms which allow the estimation and elimination of various signal distortions commonly encountered on long-distance telephone lines. The third part describes an original method of convolutional coding, adapted to this type of transmission. For appropriately chosen constellations, this method achieves low complexity by using the same trellis for different constellation sizes and by simplifying the encoding and decoding operations.

## LE COMPROMIS ENTRE BRUIT DE CODAGE ET DEBIT DE TRANSMISSION

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### SUMMARY

In most studies dealing with digital transmission at reduced bit rate and based on prediction, the system structure is not well justified: a prediction error with variance smaller than the input signal power does not necessarily yield a reduced bit rate. Depending on the location of the quantizer, the coding noise may even be considerably increased. The real objective is to maintain the power ratio of signal to coding noise while decreasing the bit rate. In this paper we compare two kinds of structure for the coder, including a filter and a quantizer. In both cases, we evaluate the number of transmission bits required to reach a given coding noise power. This confirms the usual choice of including the quantizer in the prediction loop.

## MODULATION ET CODAGE DE CANAL POUR LA RADIODIFFUSION SONORE NUMERIQUE VERS LES MOBILES

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### SUMMARY

There is an increasing interest in the development of a new broadcasting service with a view to providing an improved sound quality on portable and mobile receivers. Digital techniques have progressed over the past few years in the areas of sound programme production as well as in the field of recording. Therefore, it becomes necessary to build a coherent system from studio to user, to offer a digital audio broadcasting service, with a sound quality equivalent to the compact disc.

The modulation and channel coding system proposed by the CCETT is called COFDM (Coded Orthogonal Frequency Division Multiplex). This system is particularly suitable for high bit rate data broadcasting in multipath radio channel with varying characteristics (mobile reception in urban environment). It allows at the present time the transmission of 16 stereophonic programmes and data channels, representing a total bit rate of 5.6 Mbit/s, in a 7 MHz bandwidth.

## CONTRIBUTION FRANCAISE A LA NORMALISATION DU CODEUR DE PAROLE DU SYSTEME PAN-EUROPEEN DE COMMUNICATION RADIO-CELLULAIRE

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### SUMMARY

In this paper, we report the French contribution in the standarisation process for the digital speech coder of the Pan-European Cellular Mobile Communication Network. This study has been jointly conducted by the CNET and IBM, and has led to the normalisation of the proposed trade-off coder. In the first part of the paper, we make a short review of the different communication systems between fixed and mobile customers, and we point out the major technical and market driven arguments which have been at the origin of the project. In the second part, we sum up the different phases of the GSM's work and we give an overview of the functional architecture of the cellular network. We describe the main components and the mechanisms for the management of the mobiles and of the communications. In the third part of the presentation, we describe the different phases of the normalisation process for the digital speech coder, while paying more attention to the French contribution. After a brief presentation of the intitial proposal, we give the results of the comparative tests with other European Contributors, as well as the rationale which led us to propose a trade-off algorithm which was finally accepted by the CEPT.

## PERFORMANCES ET DEBIT CRITIQUE D'UN DECODEUR SEQUENTIEL ESTIMATEUR DE PHASE

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### SUMMARY

A new method is described for joint phase estimation and trellis-coded data decoding on random-phase channels by using a sequential decoding stack algorithm. It is based on the *Maximum a Posteriori* criterion. An appropriate Fano metric is derived for the decoder, and an expression for the computational cutoff rate is calculated. Simulation results show good performance and small additive complexity relative to the conventional sequential decoder.

# AN OPTIMAL SWITCHING ALGORITHM FOR MULTIBEAM SATELLITE SYSTEMS WITH VARIABLE BANDWIDTH BEAMS IN THE PRESENCE OF INTERFERENCE

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## SUMMARY

In a typical satellite switched time division multiple access (SS/TDMA) system, the TDMA frame is divided into several time slots, and each slot has a switching configuration permitting a certain amount of traffic to be transmitted. The system main objective is to ensure that all the traffic can be transmitted without conflict within the TDMA frame. The satellite has a number of spot beam antennas covering geographical distributed zones and a solid state RF switch on board to provide connections between the various uplink and downlink beams. A high gain antenna on board the satellite provides various spot beam coverage. This antenna may allow interconnectivity between uplink and downlink beams, i.e., substantial power from one beam may spoil over into a neighboring zone leading to what is called zone interference. The effect of zone interference has been studied in the special case of SS/TDMA system with N uplink and N downlink beams all of equal bandwidth and with at least N transponders on board.

In this work we study the traffic scheduling problem with interfering beams in a more general SS/TDMA system. We consider a system with M uplink beams and N downlink beams, where uplink i has bandwidth $\beta$ and downbeam j has bandwidth $\alpha$. The maximum traffic which can be handled by the satellite (at any given time slot) is *assumed to be* K, $\leq$ *k* $\leq$ *min* (M,N). *The interference between zones can be characterized* by the graphs $G_U$ and $G_D$ where $G_U$ = (V,E) represents the interference on uplink beams, with vertex set V representing the M zones and edge set E containing an edge ($v_i$, vj), $v_i$, vj $\epsilon$ V if and only if zone i interferes with zone j on uplink transmissions. Similarly, $G_D$ represents the interference on the downlink beams. The traffic demand is characterized by the traffic matrix D. The transmission schedule is defined as the decomposition of the traffic matrix D into several switching matrices, i.e., D = $D_1$ + $D_2$ +...+ $D_L$), where $D_i$ represents the transmissions in time slot i of the TDMA frame. The sequence ($D_1$, $D_2$,...,$D_L$) is the transmission schedule. We look for a feasible transmission schedule which never requires two interfering zones to transmit or receive simultaneously. This can be achieved if no D matrix contains entries on rows corresponding to adjacent vertices of $G_U$ or on columns corresponding to adjacent vertices of $G_D$. The proposed algorithm provides the solution of this problem and it is optimal in the sense that it minimizes the transmission time for any given traffic matrix D.

## MODEM M-FSK A SAUTS DE FREQUENCE

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### SUMMARY

This paper describes a demodulation method aimed at improving the performances of M-FSK modems using frequency hopping across a heavily crowded frequency band.

A brief theoretical study is followed by the description of a practical implementation of the proposed signal processing scheme whose performances are compared to those of a less sophisticated demodulator. The achieved processing gain increases with the severity of the jamming conditions.

## ERROR CORRECTING CODING SCHEME AGAINST PARTIAL TIME JAMMERS

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### SUMMARY

This paper proposes a new approach to anti-jamming against high energy partial time jammers concerning applications where the usual scheme spectrum spreading + error correcting coding is not efficient because of the small value of the spreading gain, due to the channel bandwidth limitation and to the minimum data rate required on the link. It consists in exploiting the short duration of the jamming pulses and trying to reconstitute the erased information by means of error correcting codes. A new method of pseudorandom interleaving, an efficient coding scheme, and a new method of erasure localization are proposed.

## DYNAMIQUE DES MÉTRIQUES DANS L'ALGORITHME DE VITERBI

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### SUMMARY

The VITERBI Algorithm is more and more implemented in systems based on optimal detection due to growing improvements in VLSI circuits. Nevertheless, for such implementation it is necessary to know the exact number of bits needed for the representation of the values computed by the algorithm. This paper describes a method to determine the true upperbound of the node metrics whose comparison gives the most likely message.

## ANALYSE DES EFFETS DE QUANTIFICATION SUR UN INTERFEROGRAMME COMPRIME

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### SUMMARY

In this communication, autoregressive modeling is used to solve a problem of data compression. The set of parameters derived from this modeling (the autoregressive coefficients, the first signal points for initialization and the model error) will be quantized, transmitted and will be used to reconstruct the signal. The effect of the quantization on each parameter is first studied. It is shown that quantization results in the introduction of several "parasitic" filters in the model. A global scheme is given, which takes into account all these effects. A complete statistical description of the errors due to the quantization of the initial conditions is carried out. Computer simulations on actual signals are given, showing the good agreement with the theorical models.

## NEW INSTANT RELATIVE DETECTION FOR MINIMUM ERROR PROBABILITY OVER NOISY BINARY-CODED CHANNELS

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### SUMMARY

The new instant relative detection is a novel approach which simultaneously minimizes the zero bit-error and the one bit-error probabilities. Sampling noise in silent communication periods allows to preset a group of thresholds corresponding to actual instant noisy conditions. Decisions are based on relative comparison of situations in the space of noise only and signal-plus-noise simultaneously. A powerful set of logical decision criteria allows decision failures in nonsure situations, which are implemented by a constant weight code for correcting undirectional errors. Achieved analytic results prove the superiority of this approach.

## SOUSTRACTION DE BRUIT PAR FILTRAGE A.R.M.A. ADAPTATIF EN TREILLIS

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### SUMMARY

Adaptive noise cancelling often applies transvesal filters to noise references. Then, a simple subtraction provides the signal estimate. Several well known methods provide good results but they are limited due to finite impulse response filter. Unfortunately increasing the width of the filter can prejudice the stabiltiy of the system.

The performances of noise cancelling are investigated when using an ARMA lattice filter. The results are generally very good but a dramatic degradation is observed when the signal to noise ratio at the input is too high.

# Topic 6
# ADAPTIVE SYSTEMS

## FILTRES EN TREILLIS ADAPTATIFS COMPARAISON DE DEUX APPROCHES DANS LE CAS DE SIGNAUX NON STATIONNAIRES

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### SUMMARY

This paper is concerned with the determination of the PARCOR coefficients of the adaptive lattice filters, in the case of nonstationary signals. Two approaches are considered. The first one is based on the optimization of the step-size of the least mean square (LMS) algorithm. The second one uses the least square algorithm with an adaptive sliding-window. These two approaches are illustrated and compared through some simulation results.

## COMPARAISON DES PERFORMANCES D'ALGORITHMES ADAPTATIFS POUR LA REJECTION DE SINUSOIDES

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### SUMMARY

In order to reject the ILS signal (composed of sinusoids) from an ILS signal with interferences, we compare the performances of adaptive algorithms used in an Adaptive Line Enhancer (ALE). We show that the variance of the ALE weights depends on the chosen algorithm: LMS or RLS algorithm; surprisingly the LMS behaves much better than the RLS. A formula for the Rejection Ratio is then given. The nature of the ILS signal imposes to use a fast RLS algorithm with a forgetting factor. We encountered the problems of such algorithms: unstable behaviours and sudden numerical divergences. A few methods to stabilize them have been tested: only one among these methods seems to be effective for our purpose.

## EFFETS DE SATURATION SUR L'ALGORITHME LMS EN ANNULATION D'ECHO AVEC DONNEES BINAIRES

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### SUMMARY

The effect of a saturation type error non-linearity in the weight update equation in LMS adaptive echo-cancellation is investigated for an independent binary data model. A nonlinear difference equation is derived for the mean norm of the difference between the estimate and the unknown filter to be estimated by the algorithm. The difference equation is evaluated numerically. It is shown that far-end binary data interference is much more deleterious to algorithm transient behavior than far-end gaussian data interference. The bit differential, for the same performance in a digital implementation of the algorithm, is studied for binary vs gaussian data as a function of the binary data power and the saturation parameter of the non-linearity.

## DEBRUITAGE PAR ANTENNE ACOUSTIQUE

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### SUMMARY

We propose an acoustical antenna which operates with a noise-only reference to cancel noise. It separately estimates and cancels the noise. Monodimensional adaptive filtering is used to estimate the noise for each sensor. The estimate of the noise is then cancelled by beamforming techniques. The frequency and temporal realizations of the antenna are detailed. Its performances are analyzed by comparison with classical techniques. It is shown that the real performance is better in the proposed antenna than in classical noise cancellation which subtracts the noise estimate. In a reverberant room, the cancellation is also more efficient than that of Frost's antenna.

## LIMITATIONS DES PERFORMANCES D'UN SOUTRACTEUR DE BRUIT DANS LE CAS DE REFERENCES POLLUEES PAR LE SIGNAL OU UN BRUIT ADDITIF INDEPENDANT

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### SUMMARY

The noise cancelling methods [Lattice, Fast-Kalman, L.M.S. or frequency algorithms] are efficient when both observation and noise reference inputs are well correlated. This statistical property is generally not verified when processing experimental signals: the reference sensor may receive a part of the signal, or an independent noise. This paper presents a study of these both cases: we calculate the residual quadratic error which depends on the Signal to Noise ratio in the reference.

## METHODES EFFICIENTES POUR L'ESTIMATION D'UN RETARD NON-STATIONNAIRE

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### SUMMARY

We propose a consistent approach to non-stationnary delay estimation in presence of noise from the observation of one signal and a noisy delayed version of itself. A first method is based on a parametric decomposition of the time varying delay on a set of K given functions. We derive THE Maximum Likelihood (ML) solution for the parameter estimation and its statistical properties. A fast algorithm is given to compute the MV parameters from the observation on a given time interval. We then demonstrate that the parametric model is equivalent to a state space model with a non linear observation equation. By linearization of this equation, a recursive solution is derived for the parameters estimates, or for the delay itself by the use of the Kalman equations. The efficiency of this approach is illustrated by simulation results in presence of fast nonstationnarity and a high level of noise.

## ADAPTIVE NOTCH FILTERING WITH HYBRID MAXIMUM LIKELIHOOD ESTIMATOR

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### SUMMARY

The conventional adaptive notch filters often fail to sense the presence of weak sinusoids or converge to the wrong frequencies. This paper proposes a new hybrid recursive (RML) and iterative quadratic maximum likelihood algorithm (IQML) to improve its performance. Extensive computer simulation results are done and reported in this paper.

## DECONVOLUTION D' UN CANAL A NON-MINIMUM DE PHASE PAR PREDICTION

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### SUMMARY

We treat the problem of restoring an unknown white sequence at the input of an unknown filter $\mathfrak{S}$ on the basis of the sequence at the filter output. It is shown that the inverse filter can be split into three cascaded filters: a backward predictor, a forward predictor and a (scalar) gain. The forward predictor is chosen as recursive in order to play the role of the decision feedback part of an equalizer in the case of data transmission. It corresponds to the minimal phase part of the filter $\mathfrak{S}$. The backward predictor is transversal and corresponds to the maximal phase part. Both predictors can be controlled in a self-adaptive mode. For telephone transmission channels, the backward predictor is more important. The learning algorithm has very low complexity and satisfactory convergence speed.

## PREDICTION DES MOUVEMENTS D'UN NAVIRE A L'AIDE DE MODELES AR OU ARMA

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### SUMMARY

This paper is concerned with the problem of k-step-ahead prediction of ship movements (rolling, pitching, heaving) by use of AR or ARMA, monovariable or multivariable models. A Monte Carlo type analysis is used to compare the performance of various predictors.

## EGALISATION ADAPTATIVE GRACE AUX FILTRES EN TREILLIS

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### SUMMARY

As we want to optimize an adaptive channel compensation system, we present here some algorithms using lattice properties. This kind of system has been already studied, but the compensation algorithm took place in the reception network. We propose a solution with the algorithm in the emission network: as we know the emission and the reception signals, we compute an estimate of the propagation channel filter using a transversal lattice estimation algorithm. Then we estimate or we compute the inverse filter using properties of lattice filters. We present two different systems and we give results for a simulation.

# DEVELOPMENT AND IMPLEMENTATION OF RLMS TIME ALGORITHMS FOR USE IN ADAPTIVE ACTIVE NOISE CONTROL

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## SUMMARY

This paper is concerned with the use of time algorithms for the adaptive digital controller needed in a single parameter active noise control (ANC) system. Considering the case of a system without feedback from the cancelling source to the noise detector, we show that a FIR filter implemented with a LMS algorithm in a home made digital controller reduces broadband noise by around 20 dB in the range studied. On the other hand, when feedback occurs between the cancelling source and the detector, there is a risk of instability. We show that an adaptive recursive filter with a RLMS algorithm can manage with this stability problem. Indeed, it achieves experimentally the same kind of attenuation simultaneously with a stable control of the feedback process. We are finally confident that these algorithms should provide interesting results in the case of multi-parameters ANC.

# Topic 7
# SONAR-RADAR

## APPORT DES FONCTIONS SPHEROIDALES POUR L'ESTIMATION DES PARAMETRES D'UNE CIBLE RADAR

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### SUMMARY

The objective of this paper shall be to show it is possible to increase discrimination of near elementary contributors from the same signal samples in using a method based on harmonic analysis. One knows that, by a Fourier transform, the normalized resolution is $1/n$ where n stands for the number of samples. The proposed method uses spheroidal windows and allows to obtain a resolution equal to $2/N$, N being the size of Fast Fourier transform.

## MODELISATION DE CIBLES SONAR, FILTRAGE A Q-CONSTANT ET TRANSFORMATION EN ONDELETTES

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### SUMMARY

Modelling a sonar target by a generalized transversal filter (set of coloured bright spots) can help in describing most of the acoustical phenomena involved in echo formation mechanisms.

In a sonar situation, this model is associated to a receive architecture based on matched filtering. Its identification leads to a set of filters matched to the echo components (integrated and differentiated replica of the emitted signal).

Due to the presence of differentiation and integration terms in the model, one needs to impose conditions on the filters responses, to make the processing feasible. The constant time-bandwidth product condition (for all the filters responses) leads to the choice of the emitted signal and to a receiver architecture based on constant-Q filtering.

We can show (for high signal to noise ratio) that the described processing can also be done using an impulse as emitted signal: instead of emitting a special signal and analyzing the echo via a set of filters matched to derivatives or integrals of this signal, one can emit an impulse and analyze the impulse response through a set of filters whose responses are integrals or derivatives of the signal autocorrelation function.

Due to the properties of the pre-defined signal, (time compression ≡ time derivative, time dilatation ≡ time integration), we show that the proposed processing is very similar to a wavelet transform.

## CLASSIFICATION DE SIGNAUX SONAR EN MODE ACTIF
## SONAR ECHOS CLASSIFICATION

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### SUMMARY

Sonar echos classification can be two folded:

— determination of geometric characteristics of a target. This is based on a discrete scatterers modelization and is carried out either with methods classically used in active sonar, or with "high resolution" methods used in spectral analysis (AR model) or in array processing ("Goniometre", TAM).

— received echos discrimination, either with linear analysis methods, or with non linear methods issued from neural networks approach.

## HAUTE RESOLUTION POUR DES SIGNAUX RADAR NON STATIONNAIRES

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### SUMMARY

Under certain conditions, a radar received signal can exhibit a very fast varying instantaneous frequency.

A precise analysis of the signal includes the search of high resolution spectral analysis algorithms for non stationary signals.

Several algorithms are examined and their resolution performances evaluated.

To begin with, we depict how the signal can be transformed into a stationary one by means of demodulation using the first order approximation of its instantaneous frequency.

Usual spectral analysis algorithms for stationary signal are then applied to the demodulated signal.

This approach is finally applied to the analysis of a non-stationary signal when the instantaneous frequency evolution rule is known and can be modeled.

## ETUDE DE BRUITS IMPULSIFS PAR ANALYSE PAR BANC DE FILTRES A Q CONSTANT

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### SUMMARY

In this paper we deal with the study of signal processing methods for the analysis of transients.

Those signals become more and more useful, in underwater acoustics, to detect and classify complex noisy sources.

The chosen method is the well known constant Q spectral analysis which has very interesting properties for characterizing such signals. We also show its analogy with the wavelets by giving its main properties.

Finally we give time-frequency results obtained on some underwater real signals.

## IDENTIFICATION AUTOMATIQUE DE BRUITS IMPULSIFS SOUS-MARINS

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### SUMMARY

Sound identification is one of the major goals of underwater acoustics. Amongst underwater sounds, signals of short duration, that we will name transients sounds, are often linked with unusual and interesting events. As these signals are nonstationnary, they are usually out of the traditional application frame of signal processing and feature extraction techniques. We present here an example of joint use of transient noises description, with autoregressive modelling and compactly supported wavelets, and significant features extraction with neural nets. This is applied to the identification of biological transient noises (context free "clics" ranging from 5 to 30 milliseconds). Correct classification rates between noises from snapping shrimp, porpoise, lobster and sea elephant range from 84% to 100% on a generalization data base containing clics not used in the learning phase.

## OPTIMUM COHERENT RADAR DETECTION IN K-DISTRIBUTED CLUTTER

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### SUMMARY

This paper deals with the synthesis of the optimum receiver for known signal in the presence of K-distributed disturbance, and with the assessment of its performance. Modelling the background noise as a Spherically Invariant Random Process yields closed form expressions for the joint pdf's of any order, thus enabling a Neyman-Pearson design of optimum detector. We show that the optimum detection amounts to processing via a zero-memory non-linearity, the distances of the received vector both from the origin and from a stored replica of the useful signal and to comparing the difference to a threshold. The optimum detector's performance assessment of the optimum detector is evaluted via computer simulations: for sake of comparison, the performance the conventional receiver under the same disturbance is also considered. An analysis of the optimum receiver operating characteristics shows that a marked improvement is achievable over the conventional receiver, at least in the region of low and moderately high detection probabilities. In particular, the larger the deviation from Gaussian distribution, the better the detectability of weak signals. The effect of the clutter correlation properties has also been investigated, as well as the influence of non-zero doppler shift of the target echo.

## QUELLES LIMITES A L'UTILISATION DU TRAITEMENT COHERENT EN SONAR HF? (100 kHz - 800 kHz)

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### SUMMARY

This paper presents some theoretical and experimental results concerning the limits set by the propagation medium to coherent processing (so-called pulse compression) in HF sonar underwater detection (100 kHz - 800 kHz). Only the limits assigned by the variations of velocity and absorption with frequency can be evaluated, theoretically, with confidence. These limits are quantifiable by dimensionless numbers involving the bandwidth B, the duration T of the acoustical pulse, the range of detection L and the features of the propagation medium. The results obtained experimentally do not call in question again the limit on B, produced by the variations of absorption with frequency, but reveal a new limit on T whose origin seems connected to non-stationarities of the medium: the sea.

## RESULTS OF A RADAR IMAGING EXPERIMENT BY ISAR TECHNIQUE

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### SUMMARY

In this paper the results of a radar imaging experiment by Inverse Synthetic Aperture Radar (ISAR) technique are presented and discussed. A one dimensional profile of an airplane in motion along a rectilinear trajectory is obtained by means of a coherent processing of the received signal collected by a conventional tracking radar.

## TRAJECTOGRAPHIE PASSIVE EN PRÉSENCE D'ERREURS DE MODÈLE: UTILISATION DES RÉSIDUS

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### SUMMARY

In the area of Target Motion Analysis and passive underwater acoustics, Batch algorithms (i.e., not time-recursive) are known to be statistically efficient and more robust than extended Kalman filters.

However, the estimation rate is usually far lower than the measurement rate, so it is necessary to test the validity of the modeling assumptions (motion with constant speed, unbiased measurements...)

By performing tests on the estimation residuals, which are analogous to innovations in Kalman filtering technics, we detect various discontinuities in the model. Parametric and non parametric tests are available and here handled for manoeuvre detection in Bearing Only Tracking: under realistic conditions (source at 10 km. changing course by 30 deg.), this manoeuvre is detected within 2 minutes.

## TRAJECTOGRAPHIE PAR COURBURE DU FRONT D'ONDE AVEC UNE ANTENNE NON ALIGNEE

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### SUMMARY

This paper deals with underwater passive target motion analysis using wave front curvature techniques (TPCFO) when time delays measurements are obtained from an array whose sensors are unperfectly aligned. Firstly, it is shown that even with a very small array misalignment, relative ignorance about immersion of the source can result in a very large bias on target range estimates. Secondly, two algorithms for estimating the position, speed, course and depth elevation angle of a target with constant velocity and depth, are presented. The first one only uses time delays measurements whereas the other uses additional measurements of the depth-elevation angle. The estimators are shown, by computer simulation, to be unbiased and statistically efficient.

## COMPARAISON DE METHODES DE TRAJECTOGRAPHIE PASSIVE EN TROIS DIMENSIONS DANS UN ENVIRONNEMENT SOUS-MARIN

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### SUMMARY

This paper deals with the passive target motion analysis in a 3D space. Several methods are defined according to the available measures and are compared by studying the observability and the potential estimation performances. The case of realistic data is investigated in order to estimate the algorithms robustness to the propagation model.

## MODELISATION DES ERREURS DES SONARS A EFFET DOPPLER

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### SUMMARY

The basic principles of Doppler effect are summarized in this description of the type of configuation adapted for measuring a carrier speed vector. A relative contribution is made for each source of error in an attempt to locate an exhaustive and realistic target. An effort is made to demonstrate their relative importance by quantifying their influence on the performance of the navigation sonar. A prototype gives an estimate of the validity of the model; the model which has been set up is proving effective for predicting the performance of the Doppler sonar.

## APPLICATION DE LA POLARIMETRIE A L'IMAGERIE ELECTROMAGNETIQUE HAUTE RESOLUTION

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### SUMMARY

Today, radar imaging is used in target rcs signatures analysis. The electromagnetic diffracting inverse problem is resolved by recovering from the knowledge of the back-scattered complex field, the geometrical coordinates and physical properties of scatterers. By calculus, we construct a High Resolution Electromagnetic Radar Image of the target in the observation plane. The Polarimetry concept, that is to say the study of the polarization, under its vectorial aspect, of the backscattered and incident waves, is introduced in radar imaging to improve target discrimination. From a complex radar target example, we show that the introduction of the Polarimetry concept in radar imaging gives an improvement in the classical problems of discrimination and classification.

## RESULTATS EXPERIMENTAUX DE L'INTERCORRELATEUR AVEC COMPENSATION DE DOPPLER DIFFERENTIEL

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### SUMMARY

This article deals with the problem of joint estimation of Time Delays and differential Doppler, and in this context provide experimental results. A motionless sensor network is used in order to localize and to estimate the trajectory of a fast moving target in the near field of the network. We recall the structure of the optimal estimator of Time delay and differential Doppler, the Time companded cross correlator; we also recall its theoretical performances, and its behaviour when several sources are present. The theoretical performances (Cramer-Rao Lower Bound) are compared with those of the actual implemented estimator. Then experimental results are provided in order to prove the good behaviour of our estimator in real and complex context.

## A COMPARISON OF MATCHED FIELD PROCESSING SCHEMES OPERATING IN THE TIME AND FREQUENCY DOMAINS

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### SUMMARY

Underwater acoustic propagation models have improved to the point where they can now be employed in matched field processing (MFP) to produce much improved source detection and localization over that obtainable on the basis of plane wave assumptions. MFP usually proceeds from the covariance matrix of signal plus noise estimated at the frequency of interest. The covariance matrix represents an averaging of spectral and cross-spectral terms after Fourier transformation of the time sequence. Stationary sources may be resolved or detected by signal processing applied to the covariance matrix. However, when an acoustic source is moving in a multimodal or multipath environment, this averaging produces a covariance matrix in which the signal spans a larger vector space and is no longer as well separated from the noise. As a result, the resolution of moving acoustic sources by orthogonal techniques will be reduced compared to that obtained for stationary sources.

This study investigates the possibility of reducing the loss in resolution and array gain of MFP that results from averaging in time to form the covariance matrix. The proposed solution is a MFP scheme operating in the time domain that retains the single frequency concept with its attendant signal-to-noise gain and modelling convenience. The signal processing algorithm resembles a Fourier transform modified by the propagation model. Performance of MFP schemes operating in the time and frequency domains are compared for moving and stationary sources.

# Topic 8
# SPATIAL PROCESSING

## LOCALISATION DE SOURCES PONCTURELLES A LARGE BANDE

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### SUMMARY

A new broad-band high resolution array processing method is presented. In order to cancel the variations of the steering vector with respect to frequency, an interpolation technique is developed, which estimates the useful signal on some synthetic sensors along the array axis. The noise correlation induced by the interpolation process is reduced by the use of a spheroidal filter bank. The method is then extended to sector analysis.

## UTILISATION DU SOUS-ESPACE SIGNAL COHERENT POUR LA LOCALISATION DE SOURCES BANDE LARGE

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### SUMMARY

Since many years, we have been interested in the study of the wide-band signals. The principal objectives are generally the detection and localization of the sources radiating these signals. Several methods using the incoherent signal subspace or the coherent signal subspace are proposed for this problem.

In this paper we propose a new transformation matrix, to focus the different estimated spectral matrices at one chosen frequency, in the analysis bandwidth. These transformed matrices are then averaged for estimating the coherent signal subspace.

## LOCALISATION DE SOURCES LARGE BANDE PAR DES METHODES TEMPORELLES

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### SUMMARY

Broadband source location methods in the time domain are based on models of the received signals. In this paper, we show the connexions between two methods corresponding to different ways of approximating the time delay effects on the source signals. In the first model, the time delays are approximated by digital filters with rational transfer function; in this case, we establish that the noise subspace $B$(f) of the spectral density matrix is generated by polynomials vectors the coefficients of which correspond to elements of the noise subspace of the spatio-temporal covariance matrix. In the second model that we borrow from the non-stationary signal domain, the vectors of $B$(f)are supposed to have a finite order expansion with respect to a basis of functions. When the basis is orthogonal, we show that the coefficients of the expansion of $B$(f) correspond to elements of the noise subspace of a covariance matrix obtained by filtering the sensors outputs. Consequently, the two methods are strongly connected.

## QUASI SOLUTIONS DE PROBLÈMES MAL POSÉS

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### SUMMARY

The characterization of independent sources contributing in a noise field can be achieved if:

– the number of measures is greater than the number of sources

– a model relating measures and sources is given (linearity, stationarity)

Our purpose is to replace a fixed model by a woolly model, nearer of real situation, and show how we can characterize it.

## ANTENNES ADAPTATIVES SOUS CONTRAINTES

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### SUMMARY

The purpose of this paper is to analyse the performance and the convergence rate of some antennas which are optimal under constraints which do not require any a priori knowledge on the desired signal. One of this contraint exhibits the concept of "power inversion" and the associated antenna is called "*Power Inversion Constraint Array*" (*PICA*). Firstly we study the performance of the antenna using the norm constraint. Then, we complete previous results by evaluating the convergence rate of the *PICA* using a Sample Matrix Inversion (*SMI)* algorithm. The relation between the *PICA* and the *Gram-Schmidt* (*GS*) preprocessor is examined and the convergence rate of these two latter structures are compared.

## UNE ALTERNATIVE AUX MÉTHODES À HAUTE RÉSOLUTION BASÉES SUR LA DÉCOMPOSITION EN ÉLÉMENTS PROPRES

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### SUMMARY

A new method is proposed for the estimation of the signal subspace from the data covariance matrix (N × N). This has two main advantages: first, it accounts for more general hypotheses on the noise, and secondly, it replaces the usual EVD decomposition by a simpler factorization of the covariance matrix, the noisy terms being excluded. The method also provides the rank r of the signal matrix, equal to the number of sources. A first initialization step takes advantage of the linear relations between rows of the matrix connected to the singularity. It assumes that the rank r is smaller than N/2. The optimization step uses a gradient algorithm with optimal step selection. It can be used in a non-stationary context to track the variations of the signal subspace.

## METHODES D'ESTIMATION SPECTRALE 2D POUR LE TRAITEMENT D'ANTENNE BANDE ETROITE

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### SUMMARY

This paper treats two high resolution 2D spectral analysis methods fitted to frequency and bearing estimates of non sinusoidal narrowband signals in array processing. The first is a 2D extension of the "goniomètre" method based on the algebraic properties of the spatio-temporal covariance matrix. The second is a generalisation of the TAM method introduced by Kung. It exploits the property that every minimal factorisation of the noise free spatio-temporal covariance matrix coincides with the observability matrix of 2D linear system realised by an Attasi model. Computer simulations are presented which compare the performances of the "goniomètre" 2D extension with the spatial covariance and interspectral matrix high resolution methods.

## METHODE LARGE-BANDE DE LOCALISATION DANS LE DOMAINE TEMPOREL

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### SUMMARY

We consider the multiple source location problem and propose an extension of the signal-subspace processing approach to the wide-band case. While most methods proposed so far are more or less straightforward extensions of the narrow-band MUSIC algorithms, our approach tries to use all the information contained in the frequency-band without splitting it into a finite number of disjoint sub-bands.

The outputs of the sensors are filtered in order to isolate the frequency-band of interest and the spatio-temporal covariance matrix of the resulting signals is estimated. This Toeplitz-bloc-Toeplitz matrix is almost rank deficient and one can thus define a signal-subspace and a noise-subspace. The method, we propose, starts by removing the contribution of the noise in that matrix and proceeds by projecting on its signal-subspace one single steering-vector which depends mainly on the direction and the isolated frequency band. It is thus typically an extension of MUSIC-like algorithm to the wideband case.

The approach is compared to existing ones on simulated examples in the case where the source-spectra are identical and flat over the isolated frequency-band. The performances appear to be satisfactory.

## MULTIPLE-SOURCE LOCALIZATION: A NEW METHOD EXPLOITING THE CYCLOSTATIONARITY PROPERTY

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### SUMMARY

A new method for detecting the number of radiating sources and for estimating their angles of location by a linear and uniform array is presented. The method exploits the property called cyclostationarity which is exhibited by all the modulated signals. The conditions of applicability of the method are: the existence and the knowledge of a cycle frequency common to each signal source, and the existence and the knowledge of a value of the lag parameter such that the cyclic cross-correlation matrix of the useful signals turns out to be full rank. The main advantage of the proposed method is its immunity to both wideband noise and narrowband interfering signals in weak signal conditions and to arbitrary and unknown interference environments.

## REDUCTION DE MODELE ET TRAITEMENT D'ANTENNE

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### SUMMARY

The aim of this paper is to present new array processing methods for passive linear arrays. The basic idea consists in modelling the sensor outputs by a linear system, then in estimating the best suited model (for a given criterion). This paper provides efficient solutions to basic problems as: approximation by a Toeplitz matrix of given rank, detection of small sources, additive noise with unknown (spatial) correlation. The presented methods represent considerable improvements with respect to the usual ones, are easy to use and inexpensive; these claims are supported with simulation results.

## LE PROPAGATEUR ET SON ESTIMATION ADAPTATIVE

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### SUMMARY

The propagator is a new concept for array processing which was first introduced under the name of "propagation operator"and since has been improved. It allows to achieve source localization and also to predict the antenna shape or to identify the complex gains of the sensors. In the present paper, a recursive-adaptive version of the propagator is given, this enables to reduce the complexity of the computation and to deal with time-varying situations like moving sources or antennas with slow varying shape. An example obtained with simulations, concerning the tracking of moving sources is shown.

## EXTENSION DES METHODES DE TRAITEMENT D'ANTENNE AU CAS DE FRONTS D'ONDES ARBITRAIRES

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### SUMMARY

Localization of impinging sources needs the knowledge of a theorical propagation model. Usually, the most used one is the plane wave model and linear array with equispaced sensors. However, in several cases in practice, discrepancies occur with the observed model. We deal in this paper with arbitrary wavefronts (unequal sensors gains and distorted phases). We present an algorithm which uses either the signal or noise subspace, or the partitioning of the cross-spectral matrix of the data. Results on underwater acoustic signals are then displayed.

# Topic 9
# SPEECH

# UNE NOUVELLE MÉTHODE D'ESTIMATION DES SPECTRES DE PUISSANCE PAR UN MODÈLE SOURCE-FILTRE APPLICATION À L'ANALYSE-SYNTHÈSE DE LA PAROLE

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## SUMMARY

The problem we consider is the approximation of a discrete power spectrum in terms of a source-filter model, and its application to analysis and synthesis of speech. We first consider the Discrete Linear Prediction [El Jaroudi 86] which does an all-pole modeling. We then present a new method where spectral envelopes are taken from an extension of the class of all-poles filter envelopes. When applying this method to real speech signals its appears that the problem has to be solved in terms of estimation: this also allows constraints on the envelopes so obtained to be taken into account. We show that our method leads to better estimations. We finally present the use of the obtained spectral envelopes for speech synthesis.

# ETUDE DES SEGMENTS TRANSITOIRES EN PAROLE A L'AIDE DE MODELES AR EVOLUTIFS ET DU CRITERE D'AKAIKE

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## SUMMARY

An automatic segmentation of the continuous speech signal, based on the Kullback divergence test, provides three different types of acoustic units:

- stationary segment
- transient segments
- short segments:

associated with three phonetic events:

- production of a target phoneme
- continuous change in articulatory configuration
- abrupt transition (closure, burst...)

The aim of this paper is to develop an algorithm which discriminates between the first and the second class (segments of the third one are easily detected, see [3]). This discrimination is based on two modelisations of the speech signal, one for each class:

- a standard autoregressive model
- a time-varying autoregressive model (the autoregressive coefficients are time dependent).

An identification of both models is carried out on each segment, the second one being done by using Yves Grenier's method; for each model, Akaike's information criterion, involving the number of parameters in the model and the likelihood (the residual error energy), is calculated in order to select the best hypothesis, i.e, the one with the lowest value of the criterion.

Experimentations have been done on phonetically balanced sentences with different implementations taking into account the following issues:

- model order
- number of functions in Grenier's model
- pre-windowing of segment
- utilization of Rissanen's criterion

The confrontation of the results with spectrograms shows the right interpretation of transitory segments: they correspond to transitions between voiced phonemes, to vocalic nuclei for which the phonetic target has been missed, and to nasal vowels.

## IDENTIFICATION DES CIBLES SPECTRALES PAR MODELES A EXCITATION MULTI-ECHELON

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### SUMMARY

Recently [1] a new model to represent the evolution of the LPC analysis has been proposed. In this approach, they are seen as the output of a vector autoregressif model excited by a piece-wise constant function. The constant parts of the excitation function are associated to spectral targets to which the output of the model tends at each instant. The jumps in the excitation function determine a segmentation of the signal and characterize each segment by the corresponding spectral target. This article extends this previous work by describing a new jump location method. We place the jumps in the excitation function by couples and we take into account the jumps already localized. The method has been applied to a speech signal from the data base BDSONS.

## INSTABILITE ET STABILITE DES ALGORITHMES DES MOINDRES CARRES TRANSVERSAUX RAPIDES EXCITES PAR LA PAROLE

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### SUMMARY

Fast least squares transversal algorithms which are numerically stable (MCRNS algorithms) with stationary inputs, diverge with non-stationary input signals like speech. We present two techniques which allow the continuous, normal operation of those algorithms for transversal adaptive filtering, when they are excited by speech signals. The first one is a reinitialization method which can be used with a large number of fast transversal algorithms; the second one combines the regularization of the forward prediction error variance and a leakage operation on the forward and backward predictors, and it leads to a particular algorithm of the Fast Transversal Filter (FTF) type.

**ADAPTIVE FILTER STRUCTURES FOR ENHANCING COCKTAIL PARTY SPEECH FROM MULTIPLE MICROPHONE RECORDINGS**

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## SUMMARY

In this paper we propose an adaptive filter structure which can significantly enhance cocktail party speech. The underlying structure is a modified "Griffiths-Jim" beamformer in which the sections are selectively adapted on the basis of a signal detection algorithm. The signal detection algorithm relies heavily on the "burst" nature of speech. The look direction is derived from crosschannel correlations and the adaptive filters are adjusted with a standard LMS procedure. The algorithms were tested in a reverberant room with a 4 microphone array. With a faraway talker and non-directional noise typical SNR improvements were 6 to 10 dB. Detailed knowledge of the geometry of the array is not required as the algorithms do not rely on perfect spacing of the sensors.

## TRAITEMENT D'ANTENNE POUR LA REDUCTION DE BRUIT SUR LA PAROLE

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### SUMMARY

In this paper, we formulate a signal estimation problem from M observations signal + noise ($s_i + b_i$) and we apply it to noisy speech signals recorded in a car. Speech signals are well correlated; as for correlation between noises, it depends on the distance between microphones.

The three methods we present are extensions of well-known methods: in each one, we create a preprocessing in order to estimate the speech signal $s_1$ from signals $s_i (i = 2..M)$. In the first method, the M outputs of the preprocessing feed an adaptive filter which is capable of responding to a signal coming from a desired direction without distortion while discriminating against noises coming from other directions (Frost's algorithm). In the second method, (M-1) reference inputs are obtained by subtracting the M outputs (of the preprocessing) between them, and the primary input is obtained by adding these M outputs. Then we compare these two methods: we compute the mean-square error in the optimal case, and give results related to the transient state when we force the final error to be identical in the 2 methods. In the third method, we allow a distortion on the speech signal to increase the noise reduction (the filter is obtained by adding an auxiliary signal to the noises).

We compare these three methods on real signals; in a first step, we suppose that the signal is the same on each microphone; in a second step, each sensor receives an observation $s_i + b_i$ and we evalute the performances for each method.

## THE INFLUENCE OF TRANSMISSION ERRORS ON THE PERFORMANCE OF A CELP SCHEME

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### SUMMARY

The influence of transmission errors on the parameters of a CELP codec for a transmission rate of 4.8 kbit/s was studied. Random as well as burst errors were considered. The sensitivity of the short-term predictor parameters, of the long-term predictor parameter, of the gain factor, and of the innovation sequence was examined. The results show that incorrect values of the long-term predictor parameters and/or the gain factor predominantly affect the speech quality.

## GAIN OPTIMAL POUR LA DEREVERBERATION DE LA PAROLE

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### SUMMARY

Using a short-time Fourier transform analysis-synthesis system, an output signal is derived from two input ones picked up by closely spaced microphones. The aim is to eliminate as much as possible the room reverberation picked-up by the microphones. The coherence function is used to segment signals and its usefullness to correct the picked-up signals magnitudes is discussed.

## APPLICATIONS DU CODAGE PREDICTIF AVEC QUANTIFICATION VECTORIELLE A LA COMPRESSION NUMERIQUE DE LA PAROLE

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### SUMMARY

Combination of vector quantization with predictive coding has been recently proposed for speech compression. The so-called CELP (Code Excited Linear Predictive Coding) technique gives high quality speech at low bit rates (4 to 8 kbps), but originally led to high complexity implementations (40 MIPS – millions of instructions per second, and 40 K RAM) incompatible with the requirements of cellular telephone applications. This is why in the past few years, much work has been conducted to reduce the CELP complexity.

This paper is organized as follows: we first remind the basic CELP techniques and we outline their native complexity. Then, we propose a new vector quantization technique which allows to reach the same quality than the original CELP, with a reduction of the implementation complexity. The proposed technique is based on one hand on a noise spectral shaping effect obtained by special pre-emphasis of the speech signal, and on the other hand on the use of a linear codebook. In the third part, we describe two applications of this technique to a full-band coder and to a base-band vocoder. We report quality tests which validate the approach.

**CODEUR CELP A EXCITATION MIXTE**

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**SUMMARY**

In most speech coders proposed nowadays for rates between 4.8 and 16 Kbit/sec, we find two filters which are respectively short term and long term predictors (LTP). The non-predictable part of the signal is approximated every frame by a sequence encoded with a minimum number of bits.

This approximation could be realized with a number of now standardized techniques: the localization of impulses (MP technique) or a more or less sophisticated search through a dictionary (CELP)... This dictionary could be either: a purely stochastic one (define a-priori) or designed from a large corpus of residuals using clustering technique (ie: LBG).

A number of variations of these algorithms have been published (for example MP followed by CELP...) but in all cases the bit rate associated with the MP, CELP and LTP coders is fixed a-priori and the order in which processing is performed is also fixed (for example LTP before MP...)

This paper demonstrates that no constraint needs to be placed on the order of processing. A unique representation can be obtained for these three coders using a "mixed" dictionary made of three parts. The first two parts (stochastic and impulse dictionaries) are constructed a-priori. The third part is modified for each frame using the excitation sequence of the preceding frames ("predictive" dictionary). The search for the vector index and the associated gain factor to be transmitted is done using a least square criteria between the original and the predicted signal (both being passed through a perceptual filter). The coder optimizes a choice between the three types of vectors (stochastic, impulse, predictive). The processing is therefore homogeneous for the different types.

## MAPPING CELP ALGORITHMS ONTO PARALLEL ARCHITECTURES

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### SUMMARY

In low rate speech coding, CELP algorithms (code excited linear prediction) produce good quality results but require high computational effort. Single-chip solutions are only possible by using several simplification techniques that cause a degradation of coder performance. Hence, for high speech quality it is necessary to use parallel processing, particularly for algorithm development and real-time applications. The structure of typical CELP algorithms is analyzed in order to investigate the efficiency of implementation on different types of parallel architectures. Some strategies for improving execution speed of the algorithms are discussed. An efficient architecture with low hardware complexity is proposed.

## HIGH-QUALITY PROSODIC MODIFICATIONS OF SPEECH USING TIME-DOMAIN OVERLAP-ADD SYNTHESIS

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### SUMMARY

A time-domain algorithm using the pitch-synchronous overlap-add (*PSOLA*) synthesis scheme has been proposed recently in the context of diphone synthesis, and it was shown to provide a very good sound quality [1]. In this paper, we analyse the reasons why the *PSOLA* synthesis scheme can be successfully applied to the speech waveform to produce high quality prosodic modifications of natural speech. The theoretical distortions brought by the algorithm are twofold: (1) a widening of the formant bandwidths for short synthesis windows; (2) a reverberation-like effect for longer windows. A practical trade-off consists of using a synthesis window twice as long as the local pitch period. In that case, the formants distortion effect is almost not perceptible. Finally, it is shown the *Time-Domain PSOLA* algorithm can also be applied to the residual excitation signal in the context of multipulse linear predictive synthesis. The spectral distortions are slightly different in this case, whereas the resulting speech quality is also judged very good.

**CODEUR A EXCITATION PAR CODE MULTI-IMPULSIONNEL: UN ALGORITHME, UN DICTIONNAIRE D'EXCITATION**

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**SUMMARY**

Analysis by Synthesis modelisation of the excitation signal for APC (Adaptive Predictive Coding) coders, has introduced successively the multi-pulse (MPLPC) and the code exciting (CELPC) coders. The CELPC coder has very interesting characteristics because it allows the production of sub- elephonic speech quality with a bit rate less than 7 kbits/s. But the drawback of this coder is its complexity which is greater than 20 million of multiplications and additions per second (x, t/s).

In this paper, we describe a new coder, called multi-pulse code exciting coder (MPCELPC). This coder combines a low bit rate, less than 8 kbits/s with a complexity less than 3 million x,t/x.

In addition, a procedure for the extraction of the exciting code book is proposed. This procedure is based on the mean-square error minimisation criteria used for coding.

A comparative study of the MPLPC, CELPC and MPCELPC coders is also presented.

# Topic 10
# MULTIDIMENSIONAL PROCESSING

## QUELQUES REMARQUES CONCERNANT L'EXTENSION DE LA MÉTHODE DE PRONY-PISARENKO AU CAS DES SIGNAUX VECTORIELS

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### SUMMARY

This communication discusses the theoretical problems arising in the extension of the "Prony-Pisarenko" identification method to the analysis of vectorial signals. The purpose of this method being the modelization of deterministic signals (generated by a linear system evolving from initial condition without input) when they are measured in the presence of an additive white noise. The communication is especially concerned by the definition of the vectorial white noise that must be used in the extension. Difficulties arising in the practical implementation of the proposed extension are mentioned.

## CONCEPTION DE FILTRES RIF 2-D OPTIMAUX POUR LA CONVERSION DE STRUCTURES D'ECHANTILLONNAGE

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### SUMMARY

The problem of conversion between different sampling structures is stated for two-dimensional signals having an orthogonal or a quincunx structure. The two types of sampling structures considered are assumed to be related with two parameters L and K which are integers. For the decimation, prefiltering followed by downsampling, and interpolation, postfiltering after downsampling, an ideal diamond shaped filter is proposed. The problem of its optimal design, in the minimax sense, is stated considering, from the one hand, infinite precision coefficients, and on the other hand, the finite wordlength case. Optimal results are presented corresponding to both types of design and also to conversion parameters with different values.

## SEPARATION D'ONDES EN SISMIQUE MARINE

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### SUMMARY

In seismic underwater research, linear arrays of equidistant sensors are frequently used. It allows to separate, in apparent velocity, the waves refracted or reflected by discontinuities between different layers of the underground and their own multiples. We adapt "high resolution" processing methods used in antenna analysis to this kind of data. These technics use spectral matrix and eigen elements properties. Different estimating ways are applied as smoothed periodogram method and "focusing". Localization and identification algorithms allow the reconstitution of each wave recorded on the array of sensors.

## PERFORMANCES DES LISSAGES SPATIO-FREQUENTIEL POUR L'IDENTIFICATION DE SOURCES CORRELEES

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### SUMMARY

It is well know that interferences destroy the performances of the high resolution methods. New estimators of the spectral matrix of the records called spatial smoothing and frequential smoothing (focusing) for the wide band signals can resolve, under certain assumptions, this problem. In this paper, we study the correlation coefficient and the conditioning of the spectral matrix in the case of two sources before and after using the spatial-frequential smoothings; thus we outline some important results.

## DETECTABILITY CONDITIONS OF HIGH RESOLUTION ALGORITHMS IN THE PRESENCE OF CORRELATED ARRIVALS

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### SUMMARY

A common and effective way of handling correlated sources in the framework of High Resolution Techniques is to use pre-processing in the form of some kind of smoothing. This paper studies bounds on the number of arbitrarily correlated sources and/or paths that can be detected using the rank of conveniently smoothed matrices. Spatial only methods, and a combination of spatial and frequency processing are considered. It is established that the number of detectable sources is a function of not only the number of sensors but also of the number of degrees of freedom present in the waveform itself. For narrow-band processing, the degrees of freedom relate to the statistical variability of the source vector, while wideband processing has further available the degrees of freedom associated with the frequency variability of the source signals.

## ESTIMATION D'UN MODELE DU BRUIT RECU SUR UNE ANTENNE DE GRANDE DIMENSION

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### SUMMARY

We present here a new method to estimate the spatial correlations of the additive noise received on a large array of sensors. The method is based on the maximization of a functional called Relative Entropy Functional and uses the outputs of beamforming as observations. After the functional definition, we propose an extension to the multi-
frequency identification. Finally, results of simulations confirm the interest of the method.

## REDUCTION DE BRUIT EN PRESENCE DE REFERENCES MULTIPLES TRAITEMENTS PARALLELE ET CASCADE

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### SUMMARY

When we have noise references, noise cancelling is similar to filtering identification. Results have already been established in the case of one reference. We propose, in this paper, to extend them for several noise references. We have considered the performance of the noise cancelling according to the architecture used:

- parallel architecture, with a simultaneous and equivalent utilisation of the references,
- cascade architecture, using successively each reference.

We have calculated, for both structures, the errors of estimation from the various parameters (signal to noise ratios, times of estimation of the filters, filter orders), when the references are previously orthogonalized. The powers of the errors are compared, allowing the choice of the best processing for a specific environment.

## 2D FIR EIGENFILTERS: A LEAST SQUARES APPROACH

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### SUMMARY

The 1D eigenfilter approach is extended for designing two dimensional FIR filters. By minimizing a quadratic measure of the error in the 2D frequency band, an eigenvector of an appropriate matrix is computed to get the filter coefficients. This method is not only simple and also optimal in the least square sense. Several numerical design examples of 2D arbitrary shape filters are illustrated to show the effectiveness of this approach.

# Topic 11
# IMAGES

## ANALYSE DES DEFORMATIONS ET DU BRUIT DANS LES TEXTURES STRUCTUREES NATURELLES

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### SUMMARY

From a structural point of view, textures are constituted by texture elements (primitives) which are repeated according to some placement rules.

The structural texture model used here assumes that the primitive spatial arrangement is regular. The texture is described by a reduced parameter set: two regularity vectors corresponding to mean distances between neighbouring primitives and a texture sample corresponding to a primitive. The above description is performed over a set of sub-images for the texture global variations modeling.

Improvements are brought to the analysis stage. On the one hand, we have completely automatized the analysis stage, on the other hand, we have extended it to the hierarchical textures. To model the local variations, a band-limited white noise can be used. We have developed an analyzer which allows to estimate the noise parameters (variance, frequency bandwidth) on the natural texture. The defects of structure in the texture are also detected by this analyzer.

## ANALYSIS AND SYNTHESIS OF QUASI-PERIODIC TEXTURES USING COMPLETE TRANSLATION INVARIANT TRANSFORM

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### SUMMARY

A model of quasi-periodic textures built on a Fourier based translation invariant transform is presented. This transform is complete in the sense that if any two signals have the same transform then they are translated versions of each other. The transform is locally applied to the texture image using a smoothing window. Feature extraction is done in the transform-space by a nonlinear averaging method. The transform space is modeled by a network of points so that each of them represents an harmonic component of the texture. Only two covariance matrices are used to characterize the variations of the periodicities. The number of parameters of the model is approximately 50. A very simple fast synthesis method is proposed and syntheses of real textures from the Brodatz album show the validity of the model.

## MODELISATION STRUCTURALE DE LA TEXTURE. EXTRACTION DU GRAIN PRIMAIRE ET DE SA REGLE DE PLACEMENT.

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### SUMMARY

Methods to extract from a texture the primary grain and to determine its rule of placement are developed. Grains are extracted by an approximation by elliptic paraboloids, their average size and their spreading about are computed as well as their placement in the image. Granulometric study of the image is then achieved. These methods are applied to pipe radiographs which verify the necessary conditions to be boolean (Poisson's repartition of grains). Influence of defects in pipes is studied.

## REPRESENTATION PYRAMIDALE PASSE-BANDE DES TEXTURES EN SYNTHESE D'IMAGES

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### SUMMARY

The use of natural textures increases a lot the realism of computer generated images. Meanwhile models of natural textures being poor, these are best described by a stamp memorized in a table. The use of this representation in computer generated images raises some problems. The solutions given in the literature are directed toward aliasing problems. This paper brings some solutions to produce quality imagery of textures at various scales.

The proposed solution uses a laplacian pyramid for the texture description. This peculiar description gives two advantages: first, it reduces the memory amount containing the texture description, and secondly it decreases the calculus cost for the reconstruction of the texture at a given scale.

The edgesmatching of non periodical texture stamps is realized by local filtering in each frequency band. Aliasing is removed by a filtering applied in only one frequency band. So, the calculus cost is kept at a low level.

The data structure of the laplacian pyramid can be extended to a multi-pyramid structure to produce quality imagery of textures with a constant level of details at any scale.

## FILTRAGE ADAPTATIF DE TEXTURE APPLICATION A LA SEGMENTATION ET A LA SYNTHESE

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### SUMMARY

In this paper we present some results in the domain of synthesis and supervised bayesian segmentation of textured images. Our approach is based on a bidimensional stochastic process X that represent the texture field, in fact a Markov Random Field (MRF). The local state of the texture field consists of a local neighbourhood vector of which we study the covariance matrix. Its features are then used in the MRF associated Gibbs distribution.

## 3D TEXTURE MODELS FOR IMAGES OF 3D SCENES

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### SUMMARY

This paper presents a *novel* and *computationally efficient* way of modelling images that result from the projective distortions of homogeneous textures laid on illuminated 3D surfaces, as they are seen by a camera. We use a "homogeneous" Gaussian Markov Random Field (GMRF) for modelling the texture. The GMRF has been shown to faithfully reproduce a vast class of micro and macro planar "homogeneous" image textures. To transform the GMRF's into 3D texture models, we reflect in the GMRF probability distribution function, the functional relationships that exist between the images of a homogeneous textured plane when viewed head-on by a camera (with either "orthographic" or "perspective" geometry), and the image produced by the same plane when it is given an orientation relative to the optical-axis of the camera. The resulting 3D texture model is a GMRF whose parameters are the texture characteristics and the surface shape, and the camera model. The model is synthesizable, and hence one can visually judge its goodness in capturing the projective distortions (seen by a camera) of a texture laid on an illuminated surface. It is simple and computationally efficient. By locally approximating 3D surfaces by planar patches, the 3D model is also suited for analytic surfaces, or surfaces which are specified by an array of surface normal directions at different points on the surface. The basic modelling concepts are also used in extracting shape information from texture. Shape parameters estimation is posed as a maximum liklihood estimation (MLE) problem.

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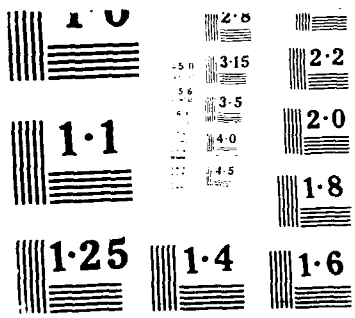

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## CODEBOOK OPTIMAL ET NOUVELLE STRATEGIE DE QUANTIFICATION VECTORIELLE D'IMAGE

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### SUMMARY

In this paper, we propose a new method of codebook design optimization combining the LBG algorithm (Linde-Buzo-Gray) and the SPLITTING technique, with its application in image coding using vector quantization.

One of the most important problems encountered in vector quantization is to find a "globally optimal" codebook. In fact, a suboptimal codebook design will always result in a suboptimal coding of the data. Now, using the LBG algorithm, the resulting codebook is a function of the initial codebook choice; the choice is very important if we don't want to get trapped in a "poor" local minimum of the distortion.

Our method modifies the initial condition of the LBG algorithm in order to go out from a local minimum of the distortion, by splitting the more frequently used codewords. So, the codebook entropy is improved (increased): a research of the best trade off between distortion and entropy is then realized.

We present, also, a new hierarchial image coding scheme using vector quantization of a residual image (original image minus the space-variant mean) in the transform domain.

## MOTION ESTIMATION USING POINTWISE MULTIGRID MEASUREMENTS

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### SUMMARY

A first and second order differential pointwise motion vector measurement is proposed and investigated so that complete motion evaluation is obtained except at singular pixels. Motion field estimation is improved by linear, non-linear filtering and multigrid recursion. Research results on natural image sequences are presented: displaced frame differences luminances show very little residual structuring thus validating the capabilities of the proposed algorithm.

# RESTAURATION D'IMAGES ET ESTIMATION DU FLOU PAR MAXIMUM DE VRAISEMBLANCE AVEC CONTRAINTES

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## SUMMARY

The problem stated in this paper is blur estimation for image restoration. In real-life applications, the characteristics of the blur are usually unknown, and the Point Spread Function (PSF) has to be estimated from the degraded image.

The identification and restoration procedure uses the fast parallel Kalman filter structure. Processing on columns is decorrelated by first applying FTT on the rows. In order to get rid of the boundary condition problems involved in this procedure, a mirror image is used. Then the 2D identification and restoration problem is transformed into a set of N 1D identification and restoration problems on the columns.

By assuming that the blur is linear and spatially invariant, the blur identification problem results in the Point Spread Function coefficients estimation. The PSF extends over a limited but large number of pixels, thus the specific problem of high number of parameters estimation is stated. Usual procedures for ARMA parameter estimation have failed in image parameter estimation, mainly due to the MA part size. The MA parameter estimator diverges on most of the scalar signal (columns).

Then parameters are constrained to be smooth by assuming that they are on a continuous function, such as a gaussian, polynomial, or wavelet function. Thus the estimation consists in hyperparameter estimation, computed by optimization of a Likelihood Function. The Maximum Likelihood technique is useful when a priori information and constraints such as energy conservation, PSF form have to be introduced. The log-likelihood function is minimized using a gradient-based procedure, with numerical computation.

The global processing consists in recursively estimating the unknown parameters using the maximum Likelihood procedure, and restoring the image using the Kalman filter. With the estimated coefficients the image is restored, and using this restored image, the parameters are optimized.

Some results on a grey-level image artifically blurred are presented.

# AUTOMATIC PARAMETER COMPUTATION FOR EDGE DETECTION BY THE ZERO-CROSSING METHOD

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## SUMMARY

Many techniques for edge detection have been proposed; among them, one of the most interesting was proposed by D. Marr and E. Hildreth. This technique is based on the zero-crossing points of the Laplacian of a Gaussian transformation of the input images, and depends on some parameters whose values must be fixed by the user. In particular, it is critical to provide the value of the threshold that selects meaningful edge points from among those extracted by the zero-crossing test. In fact this parameter exhibits very variable values that span over various orders of magnitude, depending on the image features and on the other parameters employed. A very low threshold value has the advantage of well-closed edges, but, unfortunately, also causes many false edge points due to noise. By contrast, a high threshold value allows good insensitiveness to noise, but also causes many holes on true edges.

The purpose of this paper is to present a study on the dependence of the above threshold for edge-point selection on the other parameters of the Laplacian of a Gaussian operator and on the noise level of the input image, and a criterion to compute automatically the value of such threshold. In particular, we have found that, under some hypotheses, good results can be obtained by means of a formula that depends on the noise standard deviation and on the other parameters of the Laplacian of a Gaussian operator. This formula has been derived from many experiments carried out on uniform images with zero-mean, additive, uncorrelated Gaussian noises of different variances; results have been confirmed on Magnetic Resonance images of the medical type. We are currently conducting experiments on other kinds of image.

## DETECTION DU CONTOUR EXTERIEUR D'OBJETS EN MOUVEMENT SUR FOND NON UNIFORME

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### SUMMARY

We propose a new frame difference-based operator for extraction of moving edge of objects in movement on non-uniform background and under natural light. Moving edges are first detected by combining differencing and differential operations on successive frames of the sequence. The detected boundaries are then used to generate minimum bounding octagonal models of the moving objects. The algoritihm, implemented on a standard image processor, has been tested using video-tapes of urban street crossings under various natural lighting conditions.

## MULTICRITERION DECISION MODEL AND ALGORITHM OF IMAGE RECONSTRUCTION FROM PROJECTIONS

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### SUMMARY

In this paper we propose a multicriterion decision approach to image reconstruction from projections. Our model and algorithm are applied to shepp-Logan head phantom reconstruction and computer pictures are given on VAX-11/730 microcomputer.

## RECALAGE D'IMAGES PAR ASSOCIATION DE PRIMITIVES, CONSTRUCTION ET ANALYSE D'HISTOGRAMMES MULTIDIMENSIONNELS

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### SUMMARY

A new method for image registration applied to infrared image sequences is presented. The proposed algorithm is restricted to a transformation composed of rotation and translation but it could be extended to more complex transformations. It is based on the statistical analysis of local displacements. First is presented the local displacements computation derived from the matching of feature lists. In this context, a new operator dedicated to the detection of corners and small areas in images is introduced. Then we expose the algorithm used for geometrical transformation parameters estimation. Some results on air-to-ground and ground-to-ground images are included.

## DETECTION DE MOUVEMENT PAR FILTRAGES SPATIO-TEMPORELS

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### SUMMARY

The local estimation of the main 3D orientation where spatio-temporal energy is concentrated allows to quantify motion both in direction and velocity. Three approaches founded on local filtering are compared in this paper. The first one derives the orientation from the spatio-temporal gradient. The second one consists in a linear combination of quadrature directional filters outputs. The last one is based on the computation of the optimal direction in the Fourier domain by minimization of a quadratic criterion.

## FILTRES RÉCURSIFS POUR LA DÉTECTION DE CONTOURS "LIGNES DE CRÊTES" ET "MARCHE"

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### SUMMARY

There are various kinds of edges in a single image: each of them needs a well-adapted detector. This paper presents two edge detectors: the first one concerns extracts of the "ridge-lines", whereas the second one is adapted to find closed "step edges". The use of recursive filtering improves the efficiency of the two detectors. The "ridge-lines" detector is also optimal, and the closed "step edge" detectors may be used with a multi-modal histogram and can be easily extended to a multi-scale case.

## DES PRIMITIVES AUX LETTRES — UNE METHODE STRUCTURELLE DE RECONNAISSANCE AUTOMATIQUE D'ECRITURE CURSIVE

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### SUMMARY

We present a structural method for automatic cursive script recognition. By a training step, a dictionnary and a data-base are defined. This database contains a representation of each alphabetical letter coded by means of two primitives (P1, P2) which are deduced from each drawing. In the recognition step, the input word is segmented with primitive P1. Then, P2 extraction allows to identify the letter associated with (P1, P2) coupled in database. Thus, we obtain a sequences of identified letters which define candidate words in the recorded dictionary. Finally, we retain the candidate which has the best respect to mean space factor of handwriting.

## KNOWLEDGE AND PERCEPTION: A CONTRIBUTION TO DETECT THE SHAPE OF OBJECTS IN NOISY IMAGES

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### SUMMARY

This paper describes an approach that evaluates the shape of objects in noise images by means of a set of perceptive and cognitive processes. Each pixel is labeled with a set of attributes that come from the intrinsic properties of the image, from the a-priori knowledge about the object and from reasonings about these attributes. The application of a set of rules, derived from the perception laws and from the a-priori knowledge, builds hypotheses of shape updating these attributes until the achievement of a stable label for each pixel as belonging or not to the object.

## NOUVELLES APPROCHES POUR LA SEGMENTATION ET L'IDENTIFICATION AUTOMATIQUE DES ANGIOGRAPHIES NUMERISEES

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### SUMMARY

Angiographic image analysis with a view to recognize the main vessels, requires suitable combination of digital and symbolic techniques for image processing. A thinning algorithm using distance transformation and topological properties of the pixels, allows to extract structured objects from the picture, to estimate their calibres and branchings. These objects can be grouped as sequences, taking into account calibres and deviation angles.

A heuristic approach allows to match an anatomic model describing two-dimensional configuration of the vessels, to the sequences extracted from the picture. A blackboard architecture proved to be effective for the organization of different levels of data representation, coordination of algorithm applications and implementation of analysis strategy.

## UNE FAMILLE DE DETECTEURS DE CONTOURS BASES SUR LE FILTRAGE D'ORDRE

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### SUMMARY

In this paper, a new class of gradient operators, based on order filtering, is introduced. Their properties are studied in the case of noisy images, and compared with the ones of classical operators (Sobel, Canny-Deriche). One can obtain gradient operators whose output does not depend on the edge orientation. Moreover, the coefficients of the order filters can be chosen so that it is possible to detect boundaries between areas having the same average intensity but differing by a fluctuation scale parameter.

## AN EDGE DETECTION PERFORMANCE MEASURE INCORPORATING STRUCTURAL ERRORS

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### SUMMARY

At the moment edge detection methods do not provide results acceptable for higher level image analysis methods as image segmentation and shape measurement and recognition. An edge detection performance measure tuned to these applications would contribute considerably to the evaluation of the numerous edge detection methods proposed in literature. Up to now edge detection performance measures only use simple detection error statistics and can be used on simple test images only. The new performance measure proposed here is based on the following principal features required of edge and contour images for further image processing: The edges should be complete, and without false edge points, as thin as possible, in the right position, and any clustering of errors in the detector output should be avoided.

## STRATÉGIE CONTEXTUELLE ET EXTRACTION DE PRIMITIVES POUR LA SEGMENTATION DES IMAGES MULTISPECTRALES: ETUDE D'UNE SIMULATION

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Groupe Traitement d'Images
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### SUMMARY

This paper deals with the joint utilization of spectral and spatial information in performing the segmentation of satellite multispectral images. By taking a Markovian model in a unidimensional simulation, like video scanning, we present some results by which we choose between two types of classification strategies: one using a reduced spatial context and spectral information, performed by feature extraction, another using the entire spectral information without context. The classification is based on the maximum likelihood criterion in all the cases studied. Feature extraction is realized by the Fisher discriminant approach. The simulation results show which strategy to choose in terms of spatial and spectral correlations.

## CONSTRUCTION ROBUSTE DU GRAPHE DES PRIMITIVES SEGMENTS ASSOCIEES AU SQUELETTE

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### SUMMARY

Many patterns occurring in digital image processing, like handwritten characters and many objects in automatic visual inspection, can be described by their skeletons. Practically, the skeleton must then be decomposed into segments, and structured as an attributed graph. In a real image, due to noise and structural perturbations, the graph is often strongly disturbed.

We present here a robust method for the construction of the segment primitive graph, using four rules. These rules only involve nodes and are purely local, allowing thus a good computation speed. The experimental results on real images show that the computation time is remarkably short compared with the skeletonization time. The iterative application of the rules leads to a simplification method of the skeleton, which can compete with the methods dealing with smoothed contours.

## UTILISATION DE PARAMETRES DE TEXTURES POUR REALISER L'ADAPTATIVITE D'UN SYSTEME DE CODAGE M.I.C.D.


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### SUMMARY

The work presented here is the continuation of a study made on textures in images. It's the application of found results to the building of a new DPCM Coding scheme for colour television broadcast at 34Mbit/s rate.

After a recall of the different results obtained during texture study, which is divided into two parts, an experimental one (to measure differential visibility thresholds) and another one to analyze textures (in order to extract characteristic parameters well correlated to experimental measures), we use the best parameters to adapt both predictor and quantizer.

The novelty stands also in the chrominance components coding, which uses uniquely the knowledge brought by luminance component analysis, to choose the right predictor and to adapt the quantization law.

Finally, the addition of a shifting quantization process followed by a variable length coder is proving necessary to satisfy rate constraint while having excellent quality for coded-decoded image.


## THE SHAPE INFORMATION DISTRIBUTION ON A THREE-DIMENSIONAL OBJECT


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### SUMMARY

Three dimensional curves and surfaces may be characterized by their principal curvatures. A number of researchers have used this parameterization of image data for shape identification. For a continuous surface function, the principal curvatures may be derived from local computations; furthermore, it is possible to exactly reconstruct a surface given the principal curvatures at all points on the surface. For a digital depth image with limited spatial sampling and depth quantization, a three dimensional curve or surface cannot be exactly represented. In this paper, a sampling theorem for slope-limited surfaces is developed that extends differential geometry theorems to the discrete case.

Two new parameterizations of curves and surfaces are introduced: the *shape density* identifies points of high shape interest and the *shape distribution* characterizes a total curve or surface. The shape density may be used like a three dimensional gradient (edge) detector to select salient shape identification information from a range image.

## DETECTION DES OBJETS MOBILES DANS LES SCENES NATURELLES

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### SUMMARY

We discuss the problem of moving objects detection in pictures taken by means of a fixed camera. There is no restriction relative to the studied object's displacement velocity. On the other hand, eventual changes of brightness are supposed gradual. No prior knowledge of the scene, nor its image taken without any moving object in it, is available. Hence image difference, which eliminates stationary back- and foreground, needs to be followed by another operation, in order to distinguish the moving object's current position. A survey of some interesting methods is given. Accumulative algorithms allow reference frame reconstruction. Since this reference picture contains only stationary objects, moving ones can be displayed by an absolute difference between it and the current frame. But the result is available only after an analysis of numerous frames. Edge coincidence gives a result from the second frame, but disoccluded background is interpreted as "moving", even if it was visible in the beginning of the image sequence. The proposed methods attempt to conciliate the advantages of the cited two categories and to improve the detection of the coincidence edges. Thanks to an original operation, coincidence is found for gradients. Moving objects are thus displayed from the second frame. *Static edges visible at least in two consecutive images* are accumulated in order to form a historic frame of the background and, consequently, to avoid their misinterpretation in the case of disocculsion. The techniques developed for non-binary edges are extrapolated on binary real-time detected edges. Possible applications are briefly discussed.

## REALISATION OPTIQUE DE LA TRANSFORMEE DE HOUGH EN TEMPS REEL

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### SUMMARY

The Hough transform is a powerful tool for curved detection. This paper presents an optical implementation of the Hough transform with a matrix of space-variant holograms. Using this optical processor, the Hough transform of pattern of 256 by 256 pixels is calculated in real time. The system was applied to the detection of straight lines, circles and ellipses. With this processor, all three parameters of a circle and four parameters of an ellipse are extracted from a 2-D parameter domain.

# Topic 12
# SEGMENTATION AND PATTERN RECOGNITION

## EXTRACTION DE POINTS À FORTE COURBURE À PARTIR D'IMAGES RÉELLES

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### SUMMARY

This paper deals with the high curvature point extraction problem. An approach is presented that allows to extract such points from a real image through the use of very simple and efficient measure. This measure requires directional derivatives and local maxima extraction algorithms. A theoretical study on this measure is done. It shows that this measure is well adapted for the extraction of angles. Some experimental results will be shown.

## LA DETERMINATION AUTOMATIQUE DE SILHOUETTE

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### SUMMARY

This paper presents a system designed for automatical outlining of military vehicles on infrared images.

The system's originality consists of several sets of rules, each of them specialized for a particular task (image features extraction, image content analysis). The models of vehicles are described with a semantic network formalism.

The segmentation process is guided by three knowledge sources. This organization allows the use of three types of processings: edge detection, homogeneous regions extraction, fusion of both techniques.

Picture interpretation will be achieved using a method of generation and evaluation of hypotheses. This task will be able to take into account large variations in position and conditions of observation and some partial occlusions (natural masks or countermeasures).

## LOCALISATION DE POINTS CARACTERISTIQUES DANS UN DESSIN: APPLICATION A LA RECONNAISSANCE DU CHINOIS MULTIFONTE

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### SUMMARY

Here is presented a new statistical method of multifont Chinese character recognition, which can fit to any complex drawing.

Nodes and endings, extracted from the smoothed character skeleton, are the basic features of the classification. The first classification is made according to an index of complexity, defined as the weighted sum of nodes and endings. A junction between an ending and a stroke does not disturb the index value. The second level of classification is then made by comparing the coordinates of the characteristic points of the unknown form and the model. This is realized by minimizing the Euclidian distance. A similarity index, decreasing as the distance, gives an evaluation of the likeness. Experiments on a database of one thousand of Chinese characters have been done, involving the lowest complexity characters. The recognition rate exceeds 96%.

## UNE APPLICATION DE LA THEORIE DES GRAPHES A L'EXTRACTION AUTOMATIQUE DES RESEAUX DE COMMUNICATION DANS LES IMAGES DU SATELLITE SPOT

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### SUMMARY

We propose an application of the graph theory to extract the communication networks in the SPOT satellite images. The theoretical framework is presented as the search for the shortest path on a graph. A first application is derived on a semi-automatic basis. A second method is proposed, that is based on a two step processing (primary network detection, followed by a linking of the isolated edges using a generalization of the first algorithm.). It allows the automatic extraction of a major part of the network.

# NUMBER OF FEATURES FOR HANDWRITTEN CHARACTER RECOGNITION

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## SUMMARY

Definition of features for general class of images is a difficult problem. For a particular class — handwritten lower case English script — it is shown that features can be defined on an arithmetic basis. This is achieved by dividing individual characters into fixed number of equal areas and using a finite set of primitives to define a best approximation to the contour existing in that area.

# Topic 13
# MOTION ANALYSIS IN AN IMAGE SEQUENCE

## DETECTION DE ZONES EN MOUVEMENT DANS UNE SEQUENCE D'IMAGES SELON UNE APPROCHE MARKOVIENNE

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### SUMMARY

This paper addresses the problem of motion detection in an image sequence from the variations in time of the intensity distribution. As a matter of fact, the need is not limited to change detection but encompasses the recovery of the projections of moving areas in an image. Our approach is in particular distinguished by treating conjointly these two issues, according to a probabilistic formulation. A contextual spatio-temporal information is introduced through Markovian Models. We will present two labeling models, the first one called "event-based model", the second "content-based model". Experiments with real image sequences have been carried out.

## POURSUITE VIDEO D'ICONES VARIANT EN FORME ET EN COULEUR

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### SUMMARY

In this paper, we present a tracking objects system. It provides the trajectory of moving objects from a video image sequence. The object is described by icons which are small matrix of pixels representing a discriminant pattern. The tracking system is composed of several modules. The three main ones are preprocessing, tracking window opening and localization of the icon. A dynamic icon modeling allows to follow the objects even with time-varying form and color. We study a color preprocessing which transforms a color image (3 planes RGB) into a black and white image (1 plane). The transformation is a project of colors belonging to RGB-space, on an axis. In order to have the best contrast, the chosen axis includes the target color and background one. The algorithms are performed on an image workstation controlled by a micro-computer with a video tape recorder. This tracking device is used as an assistance tool to analyse video films.

# SEGMENTATION SPATIO-TEMPORELLE DE SÉQUENCES D'IMAGES

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## SUMMARY

This paper describes an approach for the segmentation of an image sequence using motion information. The spatio-temporal continuity of the image sequence is an imporant parameter for motion analysis. When the frame rate is high, the segmentation of the image sequence into spatio-temporal regions appears to be feasible.

This segmentation can be performed with a region growing algorithm. We analyze the case of small regions in details and present an algorithm for the segmentation of moving point targets in an image sequence. A spatio-temporal pyramid is used to describe the 3D neighborhood in the image sequence and the segmentation is based on the local continuity of the trajectories.

# Topic 14
# IMAGE RESTORATION AND RECONSTRUCTION

## TOMOGRAPHIE D'OBJETS AXISYMETRIQUES: REGULARISATION PAR DES CHAMPS MARKOVIENS

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### SUMMARY

The tomographic reconstruction of an axially symmetric object can be obtained from only one projection [1]; but this inverse problem is ill conditioned. We propose here a regularization introducing local information (geometry of the materials composing the object and physical characteristics in each material) based on a markovian modelization [3].

This method leads to an algorithm experimented on a test object.

## STABILITY OF MODEL AND SELECTION OF PARAMETERS WITH APPLICATIONS IN IMAGE MEASUREMENTS

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### SUMMARY

In optical microscopy, the a priori knowledge of the nature of the object to be imaged and of the transfer function of the optical system allows to improve the limit of resolution beyond classical bounds derived from the consideration of the optical transfer only. This communication presents a quantitative study of this improvement as a function of the object model and of the image noise. The method is derived from recent studies about the limit of resolution in image restoration. An application to linewidth measurement on integrated circuits is shown.

## GENERALISED TRANSFORMATIONS IN NONLINEAR IMAGE RESTORATION

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### SUMMARY

In this paper, a general algorithm is introduced for the restoration of images formed through nonlinear systems. The approach introduced, derives the so called "generalised transformation technique". The algorithm proposed can be employed for the restoration of not only the multiplicative, but also a general class of image formation systems. In order to reveal some of the properties and advantages of the generalised transformation approach, the restoration filter based on the direct application of MMSE criterion in the multiplicative noise model, is also derived. The MMSE criterion applied on either the received or the transformed image, results in a degradation of the detailed structure. The incorporation of local adaptivity in the restoration algorithm introduced, is also addressed. The formulation of a combined criterion incorporating the MMSE and the LSE criteria is proposed. Comparisons of the techniques introduced with the direct MMSE approach are presented.

## PROBABILITÉS A PRIORI DES IMAGES ET RÉSEAUX CAUSAUX BOUCLÉS

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### SUMMARY

One of the research fields in image processing is about the probabilistic modelizations. The problems are the enormous amount of possible configurations, and also the impossibility of using causal models on bidimensional images. One solution has been recently suggested of markovian field modelization, with statistical physics descriptions [German&German]. This model can be used in a simulated annealing optimization technique, although there is still a problem for intensity images. Another problem is that the energy function U(I) has the defect to be often determined in an ad hoc manner, a small change leading sometimes to bad results.

We propose here to compute this energy function from elementary statistics of the image, following in a simple way the statistical physics, thanks to an algorithm used for the determination of a priori probabilities in expert systems [Cheeseman]. This algorithm is based on the bounded maximization of the information quantity with a langrangian technique, and can work out probabilities in non-casual nets with loops. The basic statistics could be the regions sections lengths histogram, to be computed out of a standard first approximated segmentation. One application determines the energy function for $2 \times 2$ cliques from partial monodimensional probabilities of transition between two regions.

# Topic 15
# APPLICATIONS

## EVALUATION DE QUELQUES METHODES DE DECONVOLUTION EN CONTROLE NON DESTRUCTIF

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### SUMMARY

One-dimensional deconvolution methods come up against problems in nondestructive testing when the investigated medium is highly anisotropic and inhomogeneous. This is the case of austenitic steel and we explain the difficulties encountered relative to the use of ultrasonic waves and the resulting inspection constraints. We show that the time-domain approach is the best way to solve the filtering effect of the medium on the wavelet emitted by the transducer. We introduce three time-domain methods with simulated and real results, in order to define their limits and application field.

## MODELISATION ARMAX DE SIGNAUX D'ÉMISSION ACOUSTIQUE

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### SUMMARY

Classification of acoustic emission signals is based on the choice of the represenation space. One can choose a descriptive (non parametric) approach which is very dependant on the noises, or an analytic (parametric) one, based on a mathematical model. We show the good capacity for ARMAX models to represent acoustic emission burst, but the classification deduced from those parameters is unsuccessful.

## CARACTERISATION DE SIGNAUX DE CONTROLE NON DESTRUCTIF PAR L'ANALYSE MULTI-ECHELLES

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### SUMMARY

In this article, we present results obtained by the application of a wavelet transform on nondestructive testing signals. First of all, we will present the method, and then, three different examples will help us to show the limits, but also the interest of this new tool for signal processing.

## RECONSTRUCTION 3D DE DEFAUTS PAR TRAITEMENT D'IMAGES INDUSTRIELLES

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### SUMMARY

Industrial radiography is a well-known, non-destructive testing which permits to detect flaws into pipes.

In this paper we present the different steps of a 3D reconstruction method in order to extract these flaws.

The constraints for the obtaining of radiographs are so great that only a few ones are available; it is the reason why we turned our research towards an iterative reconstruction method (like A.R.T.), that we test on simulated radiographs.

The steps are:

(1) simulation of radiographs of a steel block which contains calibrated flaws.
(2) reconstruction of these flaws by using the iterative method.
(3) generalization:
- reconstruction of the flaws from radiographs of the real block,
- comparison with the simulation results.

## MISE EN CORRESPONDANCE D'IMAGES STEREO PAR PROGRAMMATION DYNAMIQUE UTILISANT LA COHERENCE INTER LIGNES

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### SUMMARY

A dynamic programming algorithm is proposed in this paper for matching stereoscopic images. The approach is based on an intra-line search at the pixel level. To maintain the continuity of the object outside epipolar lines, the gradient of the contour is used to give a strong constraint for inter-line coherence. The advantage of this method is to ensure the inter-line coherence without the combinatorial search in 3D space. A correlation step is then performed to correct matching errors, and an interpolation is done, using also dynamic programming, after testing if occulsions exist.

## COMPRESSION D'IMAGES POUR LA MISSION D'EXPLORATION PLANETAIRE PHOBOS II

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### SUMMARY

The study realized for the international mission of planetary exploration Phobos II as a part of a French-Soviet cooperation, concerns the coding of Phobos images on the space probe. The developed coding method uses a Discrete Cosine Transform applied on 16 × 16 image blocks. An adaptative thresholding allows to select the coefficients. The coefficients are uniformly quantized and coded using fixed length codes. The threshold mask is coded using a quadtree representation. Simulation results are satisfactory. The first images will be received and decoded in April 1989.

## METHODES DE DECONVOLUTION EN SISMIQUE REFLEXION MARINE A TRES HAUTE RESOLUTION PAR FAIBLE PROFONDER D'EAU

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### SUMMARY

In shallow water with very high resolution marine seismology of the continental shelf, the recorded signals come from the reflection of an acoustic signal by the different layers of the sub-surface. The source signal used in this work is a mud penetrator at 2.5 kHz which gives a high spatial and vertical resolution with slight penetration. The received signal can be seen as the convolution of the source signal with the impulse response of the geological structure. The aim of this work is the deconvolution of the received signal in order to recover this impulse response. The main problem is that the source signal is generally not known. We proceed in two steps: we identify the source signal, then, we identify the impulse response of the geologic structure. Different methods have been tested to get a model of the source signal: AR and ARMA models and homomorphic filtering. Once the source signal is estimated, this information is used to deconvolve all the reflections. Two methods have been compared: inverse AR and ARMA filters and multipulse modeling. The combination of homomorphic filtering or mixed phase ARMA models and the multipulse modeling gives satisfactory results of the arrival times of the echoes, but some uncertainties remain as to their amplitudes.

## VERS UN SYSTEME D'AIDE A L'EXPERTISE EN RADIOGRAPHIE INDUSTRIELLE: APPLICATION DE METHODES DE TRAITEMENT D'IMAGE

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### SUMMARY

In this paper we present a synthesis of a difficult problem: application of image processing methods to industrial radiography in non destructive testing.

The selection of the image processing operators is complex regarding the quality and diversity of this kind of images.

Here we emphasize the importance of the a priori knowledges when we try to select and combine different operators.

Actually "knowledge-based system" orientation is proposed to resolve this problem.

So we develop a knowledge base coded with production rules.

Ideally, in the future this system will allow expert radiography to use the many possibilities of image processing methods.

## SURVEILLANCE VIBRATOIRE DE REDUCTEURS A ENGRENAGES: APPORT DES METHODES PARAMETRIQUES DU TRAITEMENT DU SIGNAL

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### SUMMARY

This paper deals with gearboxes monitoring and diagnosis using vibration analysis.

Synthesis of classical methods available in this field is presented to exhibit advantage of parametric modelling techniques.

In particular, when the default to detect is characterized by non-stationarities in the vibration signal. This is the case of spalling in case-hardened gears which conducts quickly to tooth breakage and then early detection is essential.

Preliminary tests, on industrial test bench (300 Kwatts) show that non-stationarity detection techniques, based on level crossing of Linear Prediction Error (LPE) obtained with autoregressive (AR) modelling, are able to early detect this default.

## METHODES DE PREVISION DE L'ETAT DE MER A COURT TERM

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### SUMMARY

We explain the methods selected or developed by the CERMA in order to constitute a short-term predicting model of the sea-state based upon waves analysis at a given point. We make the assumption that it is a stationary gaussian process, at least during a brief period of time. We recommend to adopt AR or ARMA identification techniques and to give up finitely parametrized spectra. We propose to use some sequential tests to detect qualitative changes in spectral characteristics of waves. As a quantitative approach we develop some methods of deterministic extrapolation of spectra. Validations on real data are currently processed.

# APPLICATION OF SPECTRAL ESTIMATION TO LASER DOPPLER VELOCIMETRY AND SIZEMETRY

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## SUMMARY

Analysis of signals in the spectral domain has been applied to laser Doppler velocimetry and sizemetry (LDVS) to overcome the problems encountered with common counter-type processing of noisy Doppler signals. A new LDVS processor has been developed for application to spray atomization of molten metals. In this application noise is especially a problem. Rapid solidification of molten metal results in a rough particle surface, which will decrease the signal-to-noise ratio of the detected burst signals. Other noise sources may be a high particle concentration or dirty process windows. The proposed LDVS signal processor has been evaluated by analyzing artificially generated noisy burst signals both by this processor and by a countertype processor. From the measurement results the variances of the frequency and the phase difference estimator have been obtained and compared to the Cramér-Rao lower bound of the variance. The new LDVS processor has also been tested with a set of rough metal particles and the particle size distribution is compared to a distribution measured by sieving.

# Topic 16
# IMAGE CODING AND COMPRESSION

## COMPRESSION D'IMAGES PAR TRANSFORMEE EN ONDELETTE

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### SUMMARY

The purpose of this paper is to propose a new scheme for image compression:

First, we use a wavelet transform in order to obtain a set of orthonormal subclasses of images. The wavelet functions are well localized both in the space and frequency domains. The original image is decomposed on this orthonormal basis with a pyramidal algorithm architecture. This decomposition privileges horizontal, vertical and diagonal orientations. Since wavelet decomposition is based on multiresolution and privileged directions, it tries to match human vision.

The wavelet coefficients of each class are then vector quantized. A separate optimal codebook is designed for each given resolution and direction using a training sequence and a MSE distortion measurement.

Then the input vector is classified (resolution and direction) and only the appropriate subclass of the codebook is then checked using the usual MSE. Thus the computational complexity is reduced.

Finally bit allocation of each class is adjusted in order to preserve edges and shapes.

## COMPRESSION NUMERIQUE DES IMAGES ASTRONOMIQUES A L'AIDE DE TRANSFORMATIONS MORPHOLOGIQUES

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### SUMMARY

In this paper, we present a compression method of astronomical images by morphological transformations. This method is very well adapted to their textures. The compression ratio is about 1.5 times the one of contour coding method. To go further in this direction, these morphological transformations may be applied to tridimensional images, the third dimension being the intensity of pixel.

# SYSTEME DE REDUCTION DU DEBIT D'INFORMATIONS ASSOCIE A LA TRANSMISSION D'IMAGES A HAUTE DEFINITION

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## SUMMARY

This paper deals with the problem of transmission of digital information required to enable a high definition television signal to be conveyed via narrow bandwidth channels into a MAC compatible form. These Digital Assistance informations (DA) mainly consist of motion vectors used to achieve a motion compensated interpolation of the non transmitted fields in the decoder. The digital channel capacity available in a MAC/ Packet standard is limited and therefore a bit rate reduction technique is necessary. We present a technique that takes benefit of two main postulates:

- the number of different motion vectors between two fields spaced of 40 ms is not large,
- these vectors are strongly correlated in the temporal direction.

These assumptions have been checked by simulations and have led to a bit rate reduction scheme that reduces the amount of information to be transmitted. In the same way we have developed a new motion estimation algorithm that fulfills the bit rate requirement while keeping the same picture quality.

# ETUDE COMPARATIVE DES SYSTEMES DE COMPRESSION DE DONNEES SANS PERTE D'INFORMATION ET CODAGE ARITHMETIQUE

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## SUMMARY

The classical approaches to data compression without loss of information, are based on a decoupling of the source model and coding. The combination most often use is statistic modelling with Huffman coding. In this case either a fixed model is assumed (which has the advantage of single pass-coding but which is not efficient for variable sources) or a model, which is adaptable to each file, but which requires a "double-pass". The single-pass incremental parsing algorithim of Ziv-Lempel achieves high compression ratios. However, this algorithm is only efficient for very large files correlated in single dimension.

To overcome these disadvantages, the "context" algorithm, which permits an adaptive markovian modelling of the source, was presented by Rissanen in 1983. With its tree modelling and a novel organization based on the minimal description length principle, this algorithm can be applied to sources correlated in several dimensions; such as images. The coding part is implementd by a multi-level multiplication free arithmetic code, which has the advantage, with respect to "prefix" codes, of perfectly recovering the entropy, whatever the statistic, and of being completely adaptive in the case of model modification.

This article is organized in three parts: first a review of the different data compression methods, without loss of compression, then a description of the "context" algorithm and a comparative performance evaluation for different types of file. Finally, an application to video sub-band coding is mentioned.

# Topic 17
# COMPONENTS

## A METHODOLOGY FOR ASIC IMPLEMENTATION OF DIGITAL FILTERS

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### SUMMARY

In recent years the great advance in VLSI technologies and the increasing number of application fields using digital signals have created a need for new methods for developing *Application Specific Integrated Circuits* (ASICs) in the field of *Digital Signal Processing* (DSP). In this paper a novel method, in form of a *vertially sliced synthesis system,* treating these tasks is presented. The emphasis of the methodology has been put on finding suitable filter algorithm classes, on developing a special purpose architecture, on estabishing a simple, general and efficient translation of the filter algorithm into terms of hardware. Furthermore, some *Computer Aided Design* (CAD) tools have been developed during the project.

## REALIZATION OF A MONOCHIP CONVOLUTIONAL CODER — VITERBI DECODER, IN ASIC TECHNOLOGY

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### SUMMARY

The widely used convolutional code of rate 1/2 and constraint length 7 (generator polynomials 133 and 171), in association with the Viterbi decoding algorithm, has been normalized by several organizations such as INTELSAT, EUTELSAT and NASA. The data rates requirements reach several Mbps (up to 15 Mbps).

This paper describes an ASIC realization of a convolutional coder — Viterbi decoder for this code and the codes derived from it of rate 3/4 and of low rates 1/4 and 1/8.

## ASIC ARCHITECTURES FOR DIGITAL SIGNAL PROCESSING IMPLEMENTATION

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### SUMMARY

We describe a generic pipeline architecture for high-performance DSP, which has important advantages when coupled with ASIC technology. This approach is highly amenable to automation, and allows the rapid implementation of efficient, dedicated DSP machines over a wide spectrum of functional and throughput requirements.

## VLSI ARCHITECTURES FOR MOTION ESTIMATION IN IMAGE CODING

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### SUMMARY

Block-matching motion estimation using exhaustive search is the most computation intensive task in state of the art image coding algorithms, where displacement vectors are used as predictors in DPCM inter-frame coding loops; hence the interest of designing special purpose chips based on highly parallel architectures, for use in videophone, videoconference, digital TV distribution and HDTV codecs. External interface specifications for embedding such a circuit in coders are detailed. A comprehensive overview of all possible architectural solutions for the problem is given, based on high-level synthesis methods. A maximally parallel solution is presented in full detail. It offers the possibility to adapt to various external configurations and parameters with a flexible and cascadable chip, based on a semi-systolic (25 Gop/s) operative part and involving minimal control overhead.

## A 2-DIMENSIONAL 16 POINT DISCRETE COSINE TRANSFORM CHIP FOR REAL TIME VIDEO APPLICATIONS

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### SUMMARY

This paper presents a particular decomposition of the Discrete Cosine Transform algorithm (DCT), allowing regular and modular VLSI implementations. This decomposition is then used for the realization of a 2-dimensional 16 x 16 point DCT implemented on an Application Specific Integrated Circuit (ASIC) with high through-put performances suitable for real time video applications.

## UNE ANTENNE ACOUSTIQUE ADAPTATIVE ASPECTS MATERIELS ET LOGICIELS

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### SUMMARY

This paper describes an adaptive acoustic antenna designed for sound recording. We shall present three aspects of the design: specification in accordance to the applications which are audio-conferencing and hands-free telephone, description of the hardware prototype for audio-conferencing, and eventually, implementation of the algorithm which adapts the antenna.

## ARCHITECTURE MULTI PARALLÈLE
## APPLICATIONS EN TRAITEMENT D'IMAGES

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### SUMMARY

Image processing at video rate requires a very high level of computation power. This can only be achieved using a highly parallel architecture involving processors of different kinds. This paper describes a machine joining two programmable processors: an SIMD array of GAPP processors and an array processor (Zip from Mercury). The machine is dedicated to image processing by the use of especially designed I/O processors which are responsible for high speed data exchanges and processors synchronization.

## ARMOR: ARCHITECTURE MODULAIRE RECONFIGURABLE POUR LE TRAITEMENT D'UN FLOT CONTINU DE DONNÉES EN TEMPS RÉEL

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### SUMMARY

Within the particular environment of applications processing a continuous data flow in real time, the ARMOR project is aimed at making it possible for a user to define the optimal configuration of the parallel machine that will perform his application is real time and this, without having to take into account time and hardware constraints.

To this end, we have developed a designing methodology based on the use of a formal model making it possible to state the intrinsic parallelism of an application in terms of a network of modules communicating through messages. This model can then be put into a biunivocal correspondence with an architecture characterized by a recursive structure combining true parallelism and pipe-lining. It is a reconfigurable modular architecture, in which hardware modules are interconnected according to a configuration which is deduced from the application.

## DATA-DRIVEN ARCHITECTURES FOR IMAGE PROCESSING

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### SUMMARY

The NEC $\mu$PD7281 dataflowprocessor is well suited for application in a low-cost multi processor image processing system. When used in a ring structure however, the transport capacity of the ring limits the processing power of the system. This limitation is solved by using a number of processor rings in parallel. The performance of the resulting hardware is evaluated using some basic image processing algorithms. To some extent real time image processing is possible.

## AVANTAGES APPORTES PAR UNE ARCHITECTURE A FLOTS DE DONNEES DANS UN SYSTEME DE VISION ARTIFICIELLE

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### SUMMARY

Our aim here is to describe a programmable device capable of data processing from a linear 2 MHz camera, according to acquisition rhythm. It is built around a configuration of eight NEC uPD7281 processors with a data flow architecture connected together thanks to a ring data bus. The uPD7281 uses, in combination, a data flow architecture and a pipeline outline composed of programmable modules. The data flow architecture involves the simultaneous performances of several processing with a high level of parallelism. Its uses are directed towards quality control and checking tasks. However, this operator is less efficient as far as speed is concerned than wired or microprogrammed systems are but on the other hand, it has one advantage, flexibility and the ability to perform tasks which are more complex, having a cheaper cost and sufficient computation times in most cases. Lastly, we describe an application concerning the checking of 70,000 pharmaceutical capsules per hour. This study gives prominence to the different points linked to programming as well as performance of this device.

## PROPOSITION D'ARCHITECTURE POUR LA RECONSTRUCTION D'IMAGES ET DE VOLUMES À PARTIR DE PROJECTIONS

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### SUMMARY

The image and volume reconstruction has been widely studied in the field of medical and industrial imaging. The analysis of these methods allows to distinguish two algorithm families: the projection algorithms and the retroprojection algorithms in various acquisition geometries. The implementation of these operations needs a high computational power. We present the result of the study about these various algorithms in order to extract a common algorithmic structure. Then, we propose a hardware implementation which will be used as a basic module within various image and volume reconstruction architectures.

## SUR L'IMPLANTATION PARALLELE ANALOGIQUE D'UN DETECTEUR OPTIMAL DE CONTOUR

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### SUMMARY

In this paper we are concerned with the parallel implementation of a step edge detector optimal under Canny's criteria. We show that optimal IIR filter can be derived from diffusion networks. Thus these filters lead to very efficient parallel implementation as analog combinatorial circuits.

## IMPLEMENTATION OF 2400 bps VOICEBAND DATA MODEM USING THE DSP TS68930

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### SUMMARY

The present paper describes the different steps followed from the simulation to the integration of a medium speed V22bis voiceband data modem on the S.G.S.-Thomson TS 68930 digital signal processor (DSP). The choice motivations for this DSP are mainly the computational performance and facilities it offers. It features a 3-bus structure, pipeline data flow, Harvard memory spaces, parallel processing and the possibility of complex computations. The first step before the implementation on the DSP, was to perform a global simulation in a high level language which permits to optimize the different parameters. To analyse the effect of finite word length and variable dynamics, simulation has been carried out. It performs in the Fortran language all processor computations, truncations and normalizations. It is shown that high performance was obtained with only one single DSP without any external hardware.

## APPLICATION D'UNE METHODE DE RECUIT SIMULE A L'IMPLANTATION DE TRAITEMENTS SONAR

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### SUMMARY

The purpose of this paper is to present some methods which enable to assist in conceiving signal processing software, and more specially in defining static processing organization to be installed in processing automatons. The major issue is to realize a tool for aided design, so as to propose quickly several suitable solutions, and to optimize the most interesting ones.

## ULTRASONIC MONITORING OF FLOAT ZONES IN GERMANIUM BARS

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### SUMMARY

This paper gives preliminary results obtained with a prototype float zone ultrasonic measurement system. This prototype produces on-line estimates for the float zone position thickness, and completeness. Results obtained to date indicate a standard deviation of less than 1 mm for the float zone position accuracy and an accurate indication of the float zone completeness.

## METHODOLOGIE DE DETERMINATION D'OPERATEURS SPECIFIQUES EXEMPLE D'APPLICATION DE TRAITEMENT D'IMAGE EN TEMPS REEL

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### SUMMARY

In many computation domains such as real-time digital signal processing or dynamic robot control, the great quantity of data to be computed in a short fixed time, is essentially the main encountered problem. So the use of sequential machines according on Von Neumann's principles, does not allow to completely satisfy the real-time constraints. An algebraic description of data flows which are representative of the algorithm structures is presented. The optimal use of parallelism is strongly attached to the algorithm writing. A strong relationship with Data Flow PETRI Nets is shown. Then we present a fast convolution operator for image processing.

## DESCRIPTION D'UN LOGICIEL D'ANALYSE TEMPS-FREQUENCE

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### SUMMARY

This paper presents an interactive sofware designed for time-frequency analysis developed on a personal computer PC/AT. It allows non-specialist users to compute and examine the time-frequency representations of numeric signals obtained with the spectrogram or the Wigner Distribution in its different versions. Furthermore, the system allows the acquisition of analogic signals (if it is used with a specialized card) and performs the management of the data basis of the numerical signals and of their associated results. Special attention has been devoted to the general presentation of menus and results. A visualization module allows to represent the results as two-dimensional or three-dimensional images on a graphic screen with 16 colours. Due to its standard configuration and to its simplicity this system can either be used for pedagogic or practical applications.

## MÉTHODES DE SIMULATION AU NIVEAU TRANSFERT DE REGISTRES DE PROCESSEURS ET MACHINES DE TRAITEMENT DU SIGNAL

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### SUMMARY

Our signal processing machines are built with signal processing processors. First we study internal architecture features of such processors, then machines architecture features. This allows a representative description. Processors are described at the register transfer level using a data flow graph. We study the efficiency of such simulators in order to execute them quickly using performance analysis.

# DESCRIPTION, ÉVALUATION, ET UTILISATION D'UN GÉNÉRATEUR D'ADRESSES GÉNÉRAL ADAPTÉ À LA FFT

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## SUMMARY

A fairly economical solution today to build SIMD signal processing machines is to design an elementary processor optimized for the desired application, and then to use off-the-shelf components to build all that concerns the sequencing (time control, address generation...). Thus one can get processing powers such that an elementary processor can achieve a complex radix 2 FFT butterfly in 100 ns. Then appears the problem of generating three non trivial addresses during this period of time, with a system which should also be performant for other DSP algorithms.

It is shown that the use of performant off-the-shelf general purpose address generators is not a good solution because they imply a rather long initialization requirement.

A simple post-generation system to be added to an address generator is then proposed to allow intensive use of the operating units. The exhibited solution supplies also an important shrinking of the program's length and a non negligible programming simplification that would ease the inclusion in a signal processing compiler. This structure however is not FFT algorithm specific: it can be used in a general purpose signal processing machine because, not only all of the address generator's features can be used, but all of the above results can be transposed to algorithms other than FFT, for which a classical solution would not suit. A parallel use of the described structure, that leads to more regular program coding, is then presented.

## 4LP – LOW LEVEL LANGUAGE FOR THE LINE PROCESSOR "SYMPATI 2"

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### SUMMARY

Image processing can take advantage of line processor structures. We have conceived a language that makes it possible to program a line process in a way that eliminates many of the problems previously found when using the nonconventional structures.

We shall briefly recall the line processor concept and give some details about the structure our two laboratories are developing in a collaboration. Then we shall present the programming language facilities and illustrate these facilities with some examples. Finally some performances from simulated results will show processing time. This machine with up to 128 Processing Elements (P.E.) has a theoretical power of 10 Mips per P.E.

## MISE EN ŒUVRE DE L'ALGORITHME FTF STABILISE SUR UN PROCESSEUR SPECIALISE 16 BITS A VIRGULE FIXE

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### SUMMARY

The present paper deals with the implementation of a stabilized version of the Fast Transversal Filter, recently (1988) presented by D.T.M. Slock et T. Kailath. We present a method for implementing this algorithm on a 16 bits word digital signal processor: the TS 68930 of SGS.Thomson. The reported results show that the implemented algorithm is numerically stable and keeps its performances of convergence and tracking.

## EXEMPLES D'UTILISATION DU LOGICIEL DE TRAITEMENT INTERACTIF MUSTIG

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### SUMMARY

Mustig is a graphical and interactive software for the analysis of signals, either mono or multidimensional. A given treatment is implemented using a graphical "language" based on concept of dependency graph. The software includes a graphical editor, simple and powerful, for building and modifying the graph. Using several examples, we show how to use the language to define the operations (dependency diagram, flux diagram, mathematical relations between functions or between vectors and matrices) and to extend the treatment to multidimensional signals.

## SFN: UN LOGICIEL D'E.A.O. DE TRAITEMENT DU SIGNAL

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### SUMMARY

Written in Turbo Pascal 5.0, SFN (Numerical Signals and Filters) is an integrated software package devoted to signal processing teaching. Structured from pedagogical criterions, it can be used for concrete and complete initiation for stationary signals and linear filters analysis as well as for filters design techniques. Enclosing algorithms among the most recent, SFN also allows experienced users to simply and quickly test methods of filtering.

## DEFINITION D'UN OPERATEUR SYSTOLIQUE POUR LE TRAITEMENT D'IMAGES INFRAROUGES

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### SUMMARY

In this paper we propose a processing operator that permits small target detection in real time in data flow pictures acquired from a vertical scan. This processing used a bidimensional derivative filter and a local noise estimation to obtain a normalized picture.

Using recurrent equations in computing, we define a computer shape based on systolic network. This network has the advantage to accept a second order separated filters family. A simulation in SIGNAL language (developed by IRISA) can verify this static network and fix data bus sizes.

## UN LOGICEL D'AIDE A L'ENSEIGNEMENT DU FILTRAGE NUMERIQUE ADAPTATIF

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### SUMMARY

The availability of personal computers having a high processing power and a large memory combined with improved visual means leads to efficient tools for education in all technical fields.

Adaptative filtering lends itself very well to that kind of approach. The theory behind is difficult to grasp and education tools are particularly helpful, since they provide examples and simulations at will on the spot.

In this paper a software is presented for illustrating the various techniques of adaptive filtering. The organization is described and the various options taken are justified. Several examples illustrate the versatility of this education tool.

# Topic 18
# PROPAGATION

## SUR LE DEPHASAGE ENTRE L'ENVELOPPE ET LA PORTEUSE EN ACOUSTIQUE SOUS-MARINE

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### SUMMARY

Experimental results on long range propagation, in submarine acoustics, of a modulated carrier frequency (from 10 Hz to 2 kHz) show the existence of a phase difference between the envelope and the carrier frequency introduced by the propagation. We explain this phenomena by the difference betweeen the phase and group velocities due to the dispersion (small but not null) of sound in submarine acoustics. Around 1 kHz this phase difference is calculated taking into account the dispersion due to the relaxation of boric acid. This allows us to present a complex modelization of the submarine acoustic channel. We state the measurement method and we emphasize the importance of this phase in submarine communications and in the control of minor constituents in the sea.

## ETUDE ACOUSTIQUE D'UN RIDEAU DE BULLES

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### SUMMARY

The acoustic propagation in a diphasic medium: water-air bubbles is a major problem in submarine underwater acoustics, within that framework one studies the response by transparence or reflexion of a screen of bubbles.

After recalling the behaviour of an acoustic wave in presence of bubbles, the coefficient of reflection and the coefficient of transmission are calculated in three different ways. According to the first method it is the reflection and transmission between two different environments that is calculated: the water and the screen of bubbles (considered as homogenous). After having calculated the number of wave of each medium, usual formulas of coefficients of reflexion and transmission are applied. According to the second method, it is a calculation of reverberation that is used: the contribution of every bubble is added and in the calculation of the intensity sent back by a bubble the softening due to the bubbles already met is taken into account. The third way of calculating the coefficient of reflexion is a simplification of the second method. In this case for the calculation of the intensity sent back by a bubble the softening due to the bubbles already met isn't taken into account. A third way of calculating the coefficient of transmission is simply to use the formula giving the softening according to the imaginary part of the number of wave.

## INFLUENCE D'UNE ONDE DE SURFACE PROGRESSIVE SUR LES MODES PROPAGATIFS D'UN GUIDE D'ONDE

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### SUMMARY

In this paper, we determine the change in the normal modes of a waveguide due to the presence of a monochromatic surface wave under conditions of free propagation (homogeneous wave equation) at low frequency. The amplitude of the surface wave is assumed to be small compared to the acoustic wavelength. A regular perturbatio . technique and a multiple-scale perturbation technique are employed to study the interaction between normal acoustic modes and the surface wave. This interaction generates two acoustic waves whose frequency and wavenumber are different from those of the unperturbed acoustic mode. If the interaction is non-resonant, the acoustic waves generated have a small amplitude and can be regarded as perturbations of the solution obtained in the absence of surface waves. If appropriate phase-matching conditions are satisifed one of the waves generated corresponds to a normal mode of the waveguide at a new reference frequency. In this case, two acoustic normal modes get coupled, resulting in a large-scale periodic exchange of energy from one mode to another.

## APPLICATION DE LA MODELISATION GEOMETRIQUE DE LA PROPAGATION ACOUSTIQUE SOUS-MARINE EN PETITS FONDS ET HAUTES FREQUENCES

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### SUMMARY

Propagation phenomena associated with the use of active sonars in shallow water are adquately described by a geometrical model of range-averaged intensity. This approach, classical for transmission losses computation, is extended here to temporal aspects of transmitted signals, particularly to reverberation levels: this modelization provides a very significant improvement upon usual "eigenrays" methods, in computation times as well as in results fiability. Another extension is proposed for evaluating the influence of the sea-surface rugosity, whose effect is to scatter the incident energy towards adjacent angular directions; this phenomenon is treated as a coupling effect between the different geometrical directions of propagation.

## CARACTERISATION DES MODES DE PROPAGATION PAR LEUR FREQUENCE SPATIALE – ESSAI D'IDENTIFICATION D'UN FOND MARIN MODELISE EN BASSIN

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### SUMMARY

We present an applied method to the detection of propagating modes in an acoustic waveguide (which is reduced scale model of shallow water propagation). The results obtained, are in relationship with the nature of the reflecting bottom. This known method may be considerd as a complement to methods already used, by its strong accuracy. It is based on Fourier analysis, and the apparent translation from the space domain to the time domain, by moving at constant speed the sensor in the sound field. The phase reference of the electrical signal taken from the source point out the wave-numbers as a suite of spatial frequencies. Then frequency analysis in relations with a numerical treatment, conducts to mode identification. Spectrums obtained quite different, and series of excitation angles allow modal characterization, for three types of the modelled ocean bottom: fluid, elastic, and stratified.

## PRISE EN COMPTE DES EFFETS DE DIFFUSION PAR LA SURFACE ET PAR LE FOND EN ACOUSTIQUE SOUS-MARINE

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### SUMMARY

Sound scattering effects at the boundaries can be taken into account in underwater sound propagation models provided a "plane wave bistatic scattering index" can be defined for the surface and the bottom for each possible combination of the incident and scattering directions. ECKART theory, based on KIRCHHOFF approximation, can be used to calculate index laws of relatively smooth surfaces. For a given direction of the incident wave, the scattering index can be split up into two parts: the specular direction and a continuous function describing the angular distribution of the incoherent energy as a function of the scattering direction. For low grazing angles, where the theory is not valid, "likely" results can be reconstructed by using a numerical extrapolation method specially designed to respect the conditions of energy conservation, reciprocity and symmetry.

Application of this bistatic scattering function in ray models allows scattering loss (or gain) coefficient, scattering function and the spatial coherence function to be computed for the paths which are reflected by the surface and/or the bottom.

In mode models, it can be used to compute the mode coupling matrix.

## PROPAGATION DE L'AUTOCORRELATION VERTICALE EN GUIDE D'ONDE ALEATOIRE DANS L'APPROXIMATION PARABOLIQUE

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### SUMMARY

An equation for the vertical coherence of a paraxial acoustic field is developed. This equation is an improved version of the equation introduced by Tatarskii and based on a Markovian propagation approximation.

## VALIDATION D'UN MODELE DE PROPAGATION ACOUSTIQUE ENTRE UNE SOURCE AERIENNE MOBILE ET UN RESEAU IMMERGE

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### SUMMARY

An acoustic propagation model describing sound transmission through a plane water interface is validated. Ths use of moving sources and media implied the resolution of a three-dimension problem. Experimental measurements over a lake have been realized and compared with our theoretical predictions.

## ETUDE STOCHASTIQUE DE LA SURFACE D'ONDE RECUE SUR UNE GRANDE ANTENNE EN ACOUSTIQUE SOUS MARINE

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### SUMMARY

In the context of the analysis of the underwater acoustic channel by using a long towed received antenna, in long range, deep water, horizontal propagation, the form of the wavefront received on the antenna has been shown to be carried out by using polychromatic emission [1,2,3] or wide band BPSK signals[4]. The main objective of this paper is to exhibit and study the fluctuation due to the medium itself. This analysis is performed from a polychromatic emission. It is a priori difficult to separate the fluctuations coming from the deformation of the flexible antenna, from the medium's, or source's fluctuations. On the other hand, there are very few experimental results about this problem. We propose here a statistical study based on the hypothesis that during the analysis time, the antenna is not deformed and we attribute the fluctuations to the medium itself. The statistical analysis concerns the medium fluctuation mean power, spatio-temporal correlation and spatial stationarity. However we have first to eliminate the mean antenna deformation and the mean source displacement.

## MODELISATION NUMERIQUE D'UN MILIEU MULTICOUCHE

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### SUMMARY

The transmission of an elastic wave at oblique incidence through a stratified medium is studied by a matrix formalism in three cases: the first one is the multilayered solid medium, the second one is the multilayered fluid medium and the third one is the medium composed with n layers including the last which is assumed to be semi-infinite. Finally, the geoacoustic sea bottom model, consisting in a layering of sediments sandwiched between a semi-infinite layer (basement) and a thick layer (water), is studied.

## MISE EN ÉVIDENCE EXPÉRIMENTALE DES ONDES LATÉRALES DANS LE CAS DE LA TRANSMISSION ACOUSTIQUE À TRAVERS LE DIOPTRE PLAN AIR-EAU

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### SUMMARY

A lot of work has been done on acoustic propagation through an inhomogeneous fluid, both for monochromatic and transient signals. These works are often theoretical and far from experiment. The aim of our study is to link theoretical and numerical results to results obtained from an experiment. We consider here the case of a monochromatic point-source. Experimentally, we show, by introduction of absorbing planes the existence and the properties of the different contributions (geometric and lateral) that intervene in the total refracted field, in accord with simulated theoretical results, with a plane interface air-water. The study of the total transmitted pressure field points out, in accordance to theoretical studies, some interference regions dependent on the frequency emitted by the source and the importance that lateral wave can play in particular cases.

## ANALYSE TEMPS FREQUENCE DE LA RESPONSE D'UNE COQUE SPHERIQUE ACOUSTIQUE

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### SUMMARY

Time frequency analysis displays simultaneously signal characteristics in the frequency and time domain. In this work, we studied the acoustical scattering by thin spherical shells. We chose the Wigner-Ville transform as an analysing tool because of its numerous theoretical properties. Many papers have been dealing with acoustical scattering and we have now a lot of experimental and theoretical results. So we can test the performances of Wigner-Ville analysis on echoes scattered by spherical shells.

## ROLE DU COEFFICIENT DE REFLEXION DANS LA PROPAGATION EN EAU PEU PROFONDE SUR FOND STRATIFIE

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### SUMMARY

Using an integral representation of the sound field pressure in a water layer overlying a horizontally stratified medium, we determine the normal modes for the two cases of a semi-infinite fluid and elastic bottom. Thereafter we study a geo-acoustic model of the ocean's sub-bottom consisting of a single fluid layer overlying a semi-infinite solid. The characteristic equation governing normal mode propagation is solved for different values of the fluid layer thickness. Then the particular conditions of RAYLEIGH and STONELEY waves generation are considered.

## ANALYSES TEMPS-FREQUENCES DE LA PROPAGATION D'ONDES DE SURFACE NON LINEAIRES

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### SUMMARY

Most of the published experimental work on water surface waves evolution used the classical Fourier transform at the basis of data processing. As well known, this technique systematically separates the dual spaces, namely the time space and the frequency space. Then, such processing appears to be not fully appropriate to better understand the wave train unstabilities due to nonlinear effects. One of the fundamental features of the unstability is the occurrence of simultaneous amplitude and phase (or frequency) modulations.

Differential Spectral Analysis (DSA), Evolutive Parametric Analysis (EPSA) and "Wavelets Transform" were extensively used during an experimental work addressed to the above cited unstability mechanisms.

The individual performances of the techniques above but also their complementarity are first specified. The accurate informations obtained about the instantaneous amplitude and frequency (or phase) make it possible to identify some of the most important physical processes during the wave trains propagation. New results are obtained on the relative effects of dispersion, non linearity and eventually energy input from the wind.

# Topic 19
# BIOMEDICAL APPLICATIONS

## MODELISATION ARMAX DE LA REPONSE REFLEXE A LONGUE LATENCE DE L'AVANT-BRAS CHEZ L'HOMME

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### SUMMARY

The muscular response from a stretched muscle (biceps or triceps) shows a first component which is probably due to a reflex reaction. The electromyogram measured may be analysed. We present, here a "black box" model defined by a linear filter whose input is an impluse. The corresponding state model is identified by the means of a minimization of a criterion J computed with the error between the model output and the signal. The method used is a conjugate gradient one with an explicit formula of the gradient of J. The model order and the initial point of the minimization algorithm are obtained from the NANKEL matrix of the signal.

Our method has been successfully used on several trials at different persons with several experimental conditions.

## ESTIMATION DES POTENTIELS ÉVOQUÉS EN TEMPS COURT: EXTENSION DES MÉTHODES À HAUTE RÉSOLUTION AUX SINUSOÏDES AMORTIES

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### SUMMARY

The Evoked Potential is at the base of a clinical examination and is still being developed. Work on this signal is on the one hand focused on attempts to isolate it from surrounding noise and, on the other hand, to translate its morphology in diagnostic terms. Here, we review the state of present knowledge both on the signal and on the noise. This gives us the opportunity to discuss approaches to isolation proposed in the literature. In the light of this discussion, we describe our proposal which, in fact, includes two distinct and complementary parts. The first permits the detection of artifacts without making any a priori hypothesis on the morphology of the signal. The second is based on a discerning parametric modelization (in the sense that the parameters correspond to those useful to the physician) of the signal and permits the isolation in the absence of artifact. Finally, we test our approach on real recordings, which allows us to conclude on its efficiency.

## COMPRESSION DE SEQUENCES D'IMAGES MEDICALES

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### SUMMARY

For a successful implementation of medical PACS, compression of medical image series must be implemented. The method developed consists in two main steps. A principal component analysis, first step of the conventional factor analysis of dynamic structures (FADS), is applied to the original dynamic series. A limited number of principal components (curves) and their associated spatial distribution (images) are computed. Then each image is transformed using a block quantized 2D-discrete cosine transform. To evaluate quantitatively the compression, the initial and reconstructed series are processed using routinely applied functional analyses (regions of interest methods, FADS).

## CARACTERISATION ET CLASSIFICATION DES IMAGES MEDICALES EN VUE D'UNE COMPRESSION OPTIMALE

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### SUMMARY

We suggest in this paper a new methodology which consists in choosing the optimal encoding algorithm by an expert system based on a characterization and classification of medical images by means of textural characteristics. This expert system is constructed by means of a "discriminant pyramid", which is based on successive discriminating Karhunen-Loeve classification.

## CLASSIFICATION TECHNIQUES FOR FEATURE EXTRACTION IN LOW RESOLUTION TOMOGRAPHIC EVOLUTIVE IMAGES: APPLICATION TO CEREBRAL BLOOD FLOW ESTIMATION

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### SUMMARY

In order to improve the performance of the instrumental variable method (IVM) in calculating regional cerebral blood flow (rCBF) using Single Photon Emission Computed Tomography (SPECT), and inert diffusible tracer such as $^{133}Xe$, we use Learning Algorithms for Multivariate Data Analysis (LAMDA) to classify the pixels of the images of local concentrations in the brain. The LAMDA method correctly distinguishes between extra and intra-cerebral pixels and recognizes in these last the contaminatation by bone and air passage artefact. We thus conclude that LAMDA methods can improve the reliability of images of CBF estimates.

## OBJECTIVE DIAGNOSIS OF LARYNGEAL PATHOLOGY USING THE WIGNER-VILLE DISTRIBUTION

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### SUMMARY

This paper presents a new possibility for application of the *Wigner-Ville distribution* (WVD) — as a tool for voice signal processing, used in objective diagnostics of laryngeal pathology. The described method for quantitative evaluation of some acoustic parameters, used in medical practice, covers two main levels of processing: *time-domain* and *time-frequency WVD* analysis. The time-domain analysis includes autocorrelation pitch frequency estimation, voiced/unvoiced decision and adaptive peak-to-peak measure of momentum frequencies ($Fo^{(i)}$) in time direction. A synchronous with $Fo^{(i)}$ *optimal time-smoothed WVD* (OTSWVD) analysis is discussed and an algorithm for the evaluation of the *degree of hoarseness* ($DH_{wv}$) from the OTSWVD spectrum, where the cross-terms are used, is proposed. Finally, the compared experimental results of $DH_{wv}$ and $DH_{sp}$ extracted from the conventional *spectrogram* are presented.

## AMELIORATION DE LA DETECTION ET DE LA LOCALISATION DE TUMEURS MALIGNES DE LA PROSTATE PAR TRAITEMENT D'IMAGES ULTRASONORES

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### SUMMARY

The prostatic tumour screening requires an ultrasound scanning. It is planned to design a computer assisted diagnostic system. To this end, new images are built from the original one. Each of these images is obtained from specific process of the original image. Statistical properties, like grey level mean or standard deviation in a moving window are used to produce these new images. Moreover, various processes allow extraction of textural informations. For example, new images result from computation of the Discrete Fourier Transform or of the Discrete Cosine Transform in a moving window. Other processes, which deal with co-occurence matrices or fractal dimension are presented. All these processes lead to a rather good detection of the tumour in a test-image.

## DETECTION DE REPONSES LAPLACIENNES A DES STIMULATIONS VISUELLES

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### SUMMARY

Even related sources can be separated by Laplacean operator computation. Classical extraction methods are based on averaging. A new model of Laplacean signal based on Gaussian curves is proposed. It manages Laplacean response extraction from raw signal and discrimination between signal and noise.

## RECONSTRUCTION TRIDIMENSIONNELLE DE RESEAUX VASCULAIRES

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### SUMMARY

This paper deals with the 3D reconstruction problem of vascular networks from 3 projections. The described method is based on trinocular vision results, and performs the global matching of "segments" (part of vessel without crossing-point). The segment matching problem is seen as the research of an optimal path through a weighted graph. Two algorithms are proposed: one is using dynamic programming and provides the best path according to the chosen criterion, the other carries out an exhaustive search of all the possible paths. Experimental results are given for this last method.

## COMPRESSION D'IMAGES RADIOLOGIQUES PAR TRANSFORMATION DISCRETE EN COSINUS

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### SUMMARY

For digital images storing and transmitting, especially in the case of radiological images, it is necessary to use in the better way the storage memory and the bandwidth of the communication chanel. In order to reduce memory requirements for medical-purpose computer equipments, and also to reduce bandwidth requirements for the transmission of these images between hospital services, it is necessary to code the digital images before storing and transmitting them. One of the so-called first generation techniques for coding [1], which has enabled to achieve good performances in coding radiological images, is based on the two-dimensional discrete orthogonal transform utilization. In this paper, we present results of simulations for coding such images using the two-dimensional Discrete Cosine Transform (2D-DCT). An empirical expression based bit-allocation table [2], has been used to code the quantized transformed image. The original digital image represented by an $n \times n$ matrix, is divided into $m \times m$ subimages. In this work, $n = 256$ and simulations have been done with $m = 16$, $m = 32$ and $m = 64$. It is shown that the quality reconstruction improves as m increases. Among several images, the maximum compression ratio obtained, keeping a good image quality, was 12:1 with $m = 64$, 6:1 with $m = 32$ and 4:1 with $m = 16$. With post-processing after decoding, a compression ratio of 16:1 with $m = 64$ is going to be achieved soon.

## SURVEILLANCE OBSTETRICALE UTILISANT LE SIGNAL ELECTROMYOGRAPHIQUE UTERIN

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### SUMMARY

The uterine electromyogram (Electrohysterogram, EHG), is recorded by means of biopolar abdominal electrodes on women pertaining to two different clinical classes: the first one contains efficient contractions (parturition), the second one contains inefficient contractions (pregnancy or failed induced labour).

A first discriminant analysis performed on the relative energies computed from different frequency bands of the Power Density Spectrum, permits us to select the bands which separate the two classes. A set of three parameters (relative energies in the selected bands and duration of the contraction) is defined. A second discriminant analysis, using the a priori knowledge of the classes, is then performed to demonstrate that the parameter set is characteristic of the differences between the classes.

A single criterion, deduced from this analysis, provides the "efficiency" information.

This result leads to the realization of an obstetrical monitoring device for EHG analysis: the monitor is used for parturition control as a complement of the classical systems and gives important information to the clinicians about uterine contraction efficiency.

## ANALYSE ET DECISION EN ELECTROMYOGRAPHIE

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### SUMMARY

A description of the diagnosis framework designed for neurological information processing, in particular the electromyographic signal (EMG), is reported through a synthetic review of the most recent works. Emphasis is given on integration of representation methods, statistical analysis and understanding techniques. Knowledge-based approaches can be used to model and to conduct the overall resolution process. They can also make easier the low-level steps for the recognition of elementary waveforms and all their possible superimpositions.

## THE PRACTICAL APPROACH TO DETECTION OF EVOKED RESPONSES

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### SUMMARY

A method of detection of disturbances in nonstationary signal of finite time duration has been considered in the paper. The problem is very important in medical research, in the analysis of evoked potential signals. In order to formulate a useful method it is necessary to accept a difficult assumption that no statistical characteristics are known a priori and the amount of the processed data is relatively small. The concept of the solution of the problem lies in creating a discriminant vector using signal values at selected time instants. A special recursive routine was found for the points selection and discriminant vector evaluation. The routine, presented in the paper, is optimized for minimum calculation time. Additionally, a concept of the detection threshold evaluation is given. The method is illustrated with some results obtained from actual ECoG signal analysis.

## RECONSTRUCTION D'IMAGE 3D EN ANGIOGRAPHIE NUMERIQUE ET A PARTIR DE DEUX PROJECTIONS ORTHOGONALES

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### SUMMARY

In Digital Subtraction Angiography, 3D reconstruction from two orthogonal projections can be reduced into a 2D reconstruction of parallel binary-patterns cross sections. After a short review of some currently developed methods, two new algorithms and experimental results are presented. The first one is based on an equal divisor curve of each section. The second one uses a binary matrix model for the first section and yields optimal solutions with respect to the models by a minimum cost capacitated network flow.

# INVITED PAPERS

## HIGH RESOLUTION PASSIVE ARRAY PROCESSING: AN OVERVIEW OF PRINCIPLES

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### SUMMARY

High resolution array processing has been an important field of interest. This is due to the fact that often, and this is particulary true for passive sonar, the array aperture is small, which results in a weak resolving power, and also secondary lobes are difficult to maintain at a sufficiently low-level to reduce the masking effect due to high level jammers.

This paper presents an overview of the principles of high resolution passive array processing and of their properties. Performance bounds of array processing are examined, which show that resolving power can be enhanced by increasing the observation time. Resolving power of high resolution methods has the same behaviour, but with a lower gain. High resolution methods beamforming needs an estimate of the number of sources: a unified method is presented which simplifies this problem. Extension to wideband signals is presented and results obtained on real data are shown.

## PERFORMANCE LIMITS FOR HIGH RESOLUTION SYSTEMS

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### SUMMARY

This paper obtains Cramer-Rao bounds on the average location and separation of closely spaced sources. It explores the dependence of these bounds on array geometry, signal to noise ratio, and other practically relevant parameters.

## HIGH RESOLUTION PROCESSING TECHNIQUES FOR TEMPORAL AND SPATIAL SIGNALS

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### SUMMARY

The first modern generation of signal processing algorithms used Fourier methods. A second class of signal analysis and parameter estimation methods using a general linear model includes the linear predictive and signal subspace methods for parameter estimation. A third class of signal processing methods based on time-frequency analysis of nonstationary signals began with the work of Wigner (1932) and Ville (1948) and has seen extensive theoretical development and the beginning of hardware implementation. Recent work has combined linear predictive modelling with Wigner-Ville analysis. The near future will see the combination of time-frequency distribution concepts with eigensystem-based signal analysis, wideband ambiguity functions, and wideband Wigner-Ville analysis in which Doppler is treated as a time compression or expansion.

## UTILISATION PRATIQUE DE METHODES PARAMETRIQUES EN TRAITEMENT D'ANTENNE

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### SUMMARY

Practical utilization of parametric methods for array processing is not without difficiulties; among them one may mention: lack of knowledge of spatial noise correlation, number of sources, etc. Practical utilization of these methods requires however a satisfying robustness with respect to these hypotheses, leading to the definition of original methods whose practical usefulness will be proved.

## WAVELETS AND SIGNAL OR IMAGE CODING

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Wavelets expansions provide decompositions of signals and images into series of terms which are (1) self-similar, (2) both sharply localized in position and frequencies and (3) the coefficients of the wavelets appearing in the expansion are expected to yield numerical compression. These coefficients are easily computed if the wavelets expansion is orthogonal.

Such decompositions seem adapted to transient signals analysis and I first would like to report on the most significant results which will be unveiled during the conference "Wavelets and Applications", taking place in Marseille-Luminy, May 29–June 3, 1989.

I would also like to show that wavelets expansions should look quite natural to people acquainted with signal and image coding since they are implicit in most of the work on *quadrature mirror filters*.

## LA COMPREHENSION DE SCENES DYNAMIQUES PAR L'ANALYSE DE SEQUENCES D'IMAGES

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### SUMMARY

After a brief historical summary of image sequence processing, we propose a general approach to construct interpretation schemes of dynamic scenes. For a large class of applications, this aim needs efficient conceptual and algorithmic steps concerning scene and motion modelling, "low-level" primitives extraction and interpretation by spatiotemporal labelling in a more semantic sense.

Such dynamic scene interpretation schemes have been presently studied and in section II we propose experiments which have been carried out with real broadcast image sequences for coding purpose. Some results on the different algorithmic steps (segmentation, optical flow estimation, interpretation...) are illustrated.

**LES RESEAUX DE NEURONES**
**SITUATION ET PERSPECTIVES**

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**SUMMARY**

This paper is an elementary introduction to networks of formal neurons; it endeavours to give a cursory presentation of the state of the art in basic research and applications. In a first part, we describe the usual models of formal neurons and the network architectures which are currently used: static (feedforward) nets and dynamic (feedback) nets. In the second part, we give an overview of the main potential applications of neural networks: pattern recognition (vision, speech), signal processing, automatic control. Finally, the main realizations (simulation software packages, special-purpose simulation machines, integrated circuits) are outlined.

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